Houyuan VoIP PBX User Manual

HOUYUAN® IPPBX

Product Guide

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Content

Contact Houyuan

The Introduction of Houyuan

Founded in 2009, Houyuan technology has been always endeavoring in the R&D and manufacturing of the internet communication terminals. The product line of Houyuan includes IP Phone,IP PBX,POE Switches,Wireless Router,Asterisk Card...

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Chapter 1 the Introduction of IP0x

Overview of the IP0x

The IP0x is a complete Asterisk Appliance with one dual port FXO or FXS module. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT features. It provides a solid, uniform platform for traditional PSTN communications as well as VoIP communications.

Targeting for SOHO user and SMB market with an easy to use graphical interface, IP0x provides a cost-saving solution on their telecommunication/data needs. With IP0x, company with branch offices in different countries can be easily combined together to work like a virtual single office through Internet.

Features

Open Source Asterisk IP PBX
High performance OSLEC (Open Source Line Echo Canceller)
Configurable IVR menu
Voice Mail, Voicemail to Email
Call forward, Call waiting, Call transfer
Call conference
Call queues, Ring group
SIP trunk, IAX trunk, PSTN analog trunk
Call Detail Record
Access via: SSH/telnet/web
Firmware upgradable via web page
50+ available SIP/IAX2 extensions
20 concurrent calls

Applications

SOHO/SMB telephony system
Hosted service
FAX terminal
IVR system

Interface

1&2*RJ45 port
1 * Power port
2 -8* RJ11 port (FXS/FXO interchangeable)
1-4*Dual port FXO/FXS module slot

Hardware

CPU: 400MHz Blackfin 532 Chip
1-8 analog (FXO/FXS) module interface

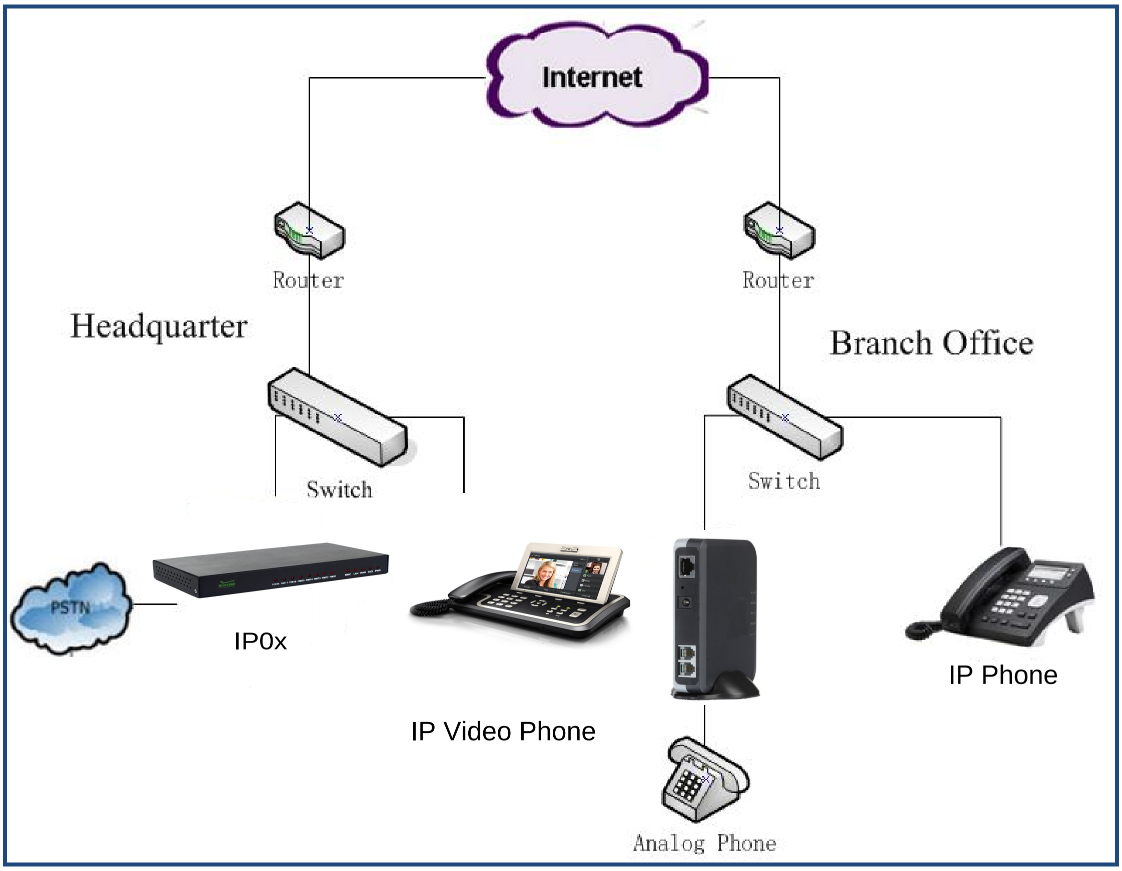
NAND flash 256 MB

SDRAM 64MB

System

Open Source uClinux

For the usage of IP0x in VoIP field, you can refer to the following network topology.



Chapter 2 Access to the IP0x

You need a PC to access to the IP0x, there are four ways for you to access the IP0x:

1. Web page access by browser
2. SSH access by putty
3. Access by browser with Fallback IP Address
4. Console port access by RS232 console cable

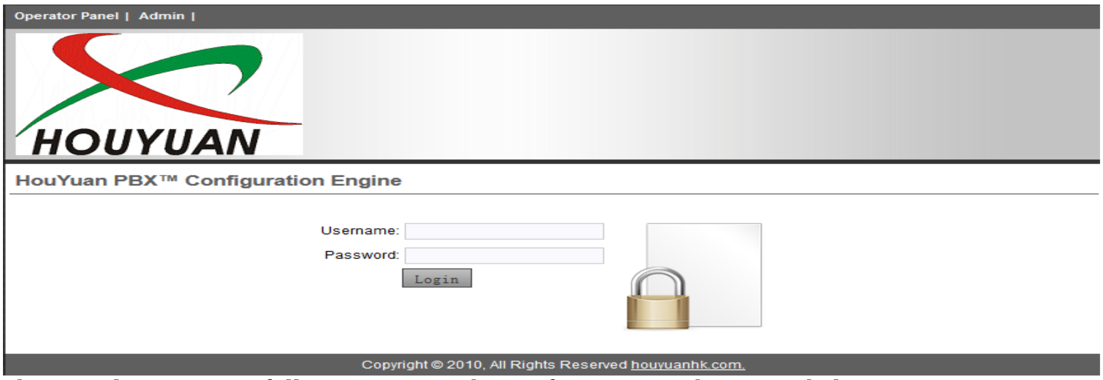
In order to access to IP0x by the first three ways, you have to check that if your network connection between IP0x and PC is OK. If you do not have network connection between IP0x and PC, you can try to use the last way to access to IP0x and change the IP address for IP0x.

2.1 Web Page Access by Browser

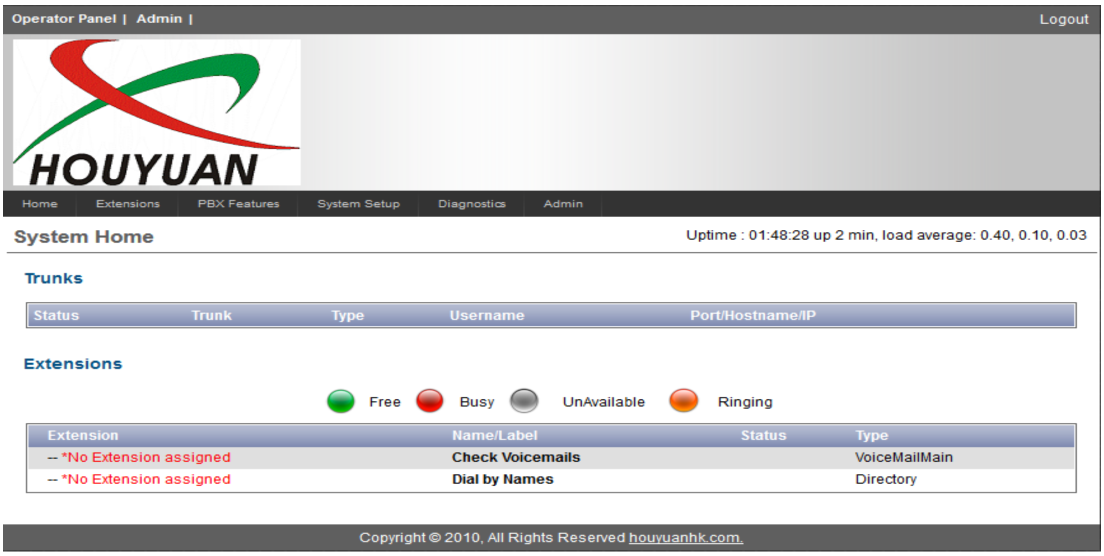
It is the most convenient and common way to access the IP0x, you just need to open your browser and input the IP address of IP0x WAN port (the default IP address is 192.168.1.100). You would better use Firefox instead of IE, because there are compatible issues.

Then input the default Username: admin; Password: admin

in the presented screen like the following:



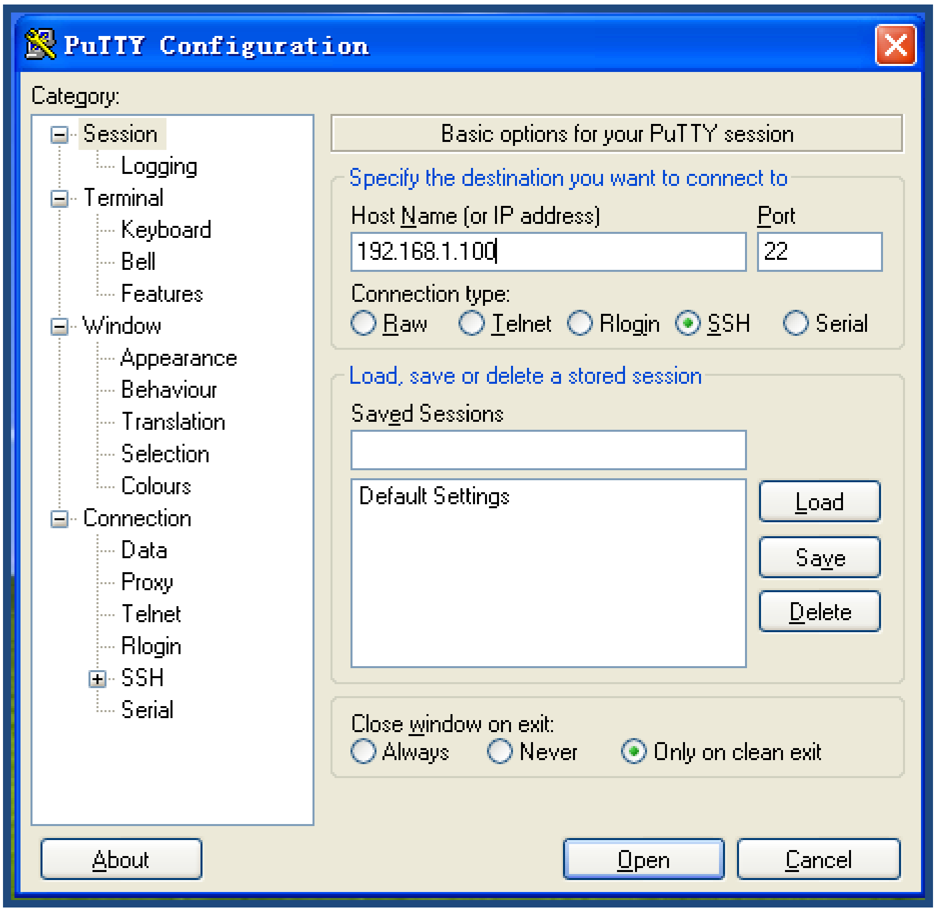
When you login successfully, you can get the configuration web page as below:



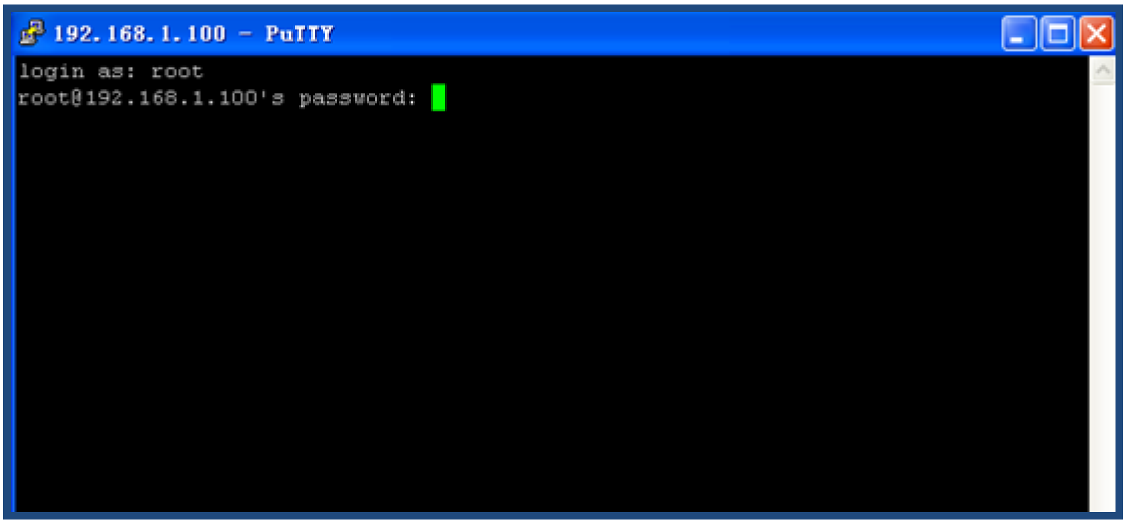
2.2 SSH Access by Putty

Logging into IP0x by SSH, you can configure IP0x by Linux command.

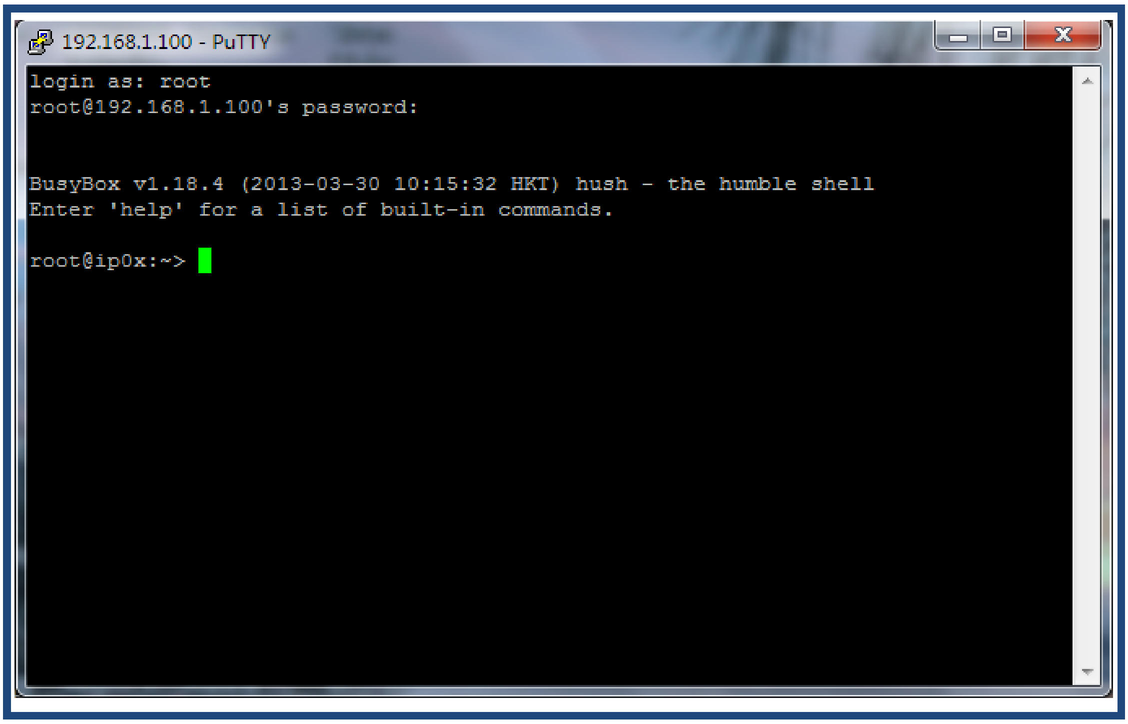
1) Please open your putty software, and input the IP0x IP address in the **Host Name** textbox, input port number in the **Port** textbox, click the **SSH** Connection type, then click **open** button. Please refer to the following screen:



2) Please input username: root, and the default password: uClinux in the following screen, you can access to IP0x successfully.



When you log into IP0x successfully, you can get the following illustration:



2.3 Access by Browser with Fallback IP Address

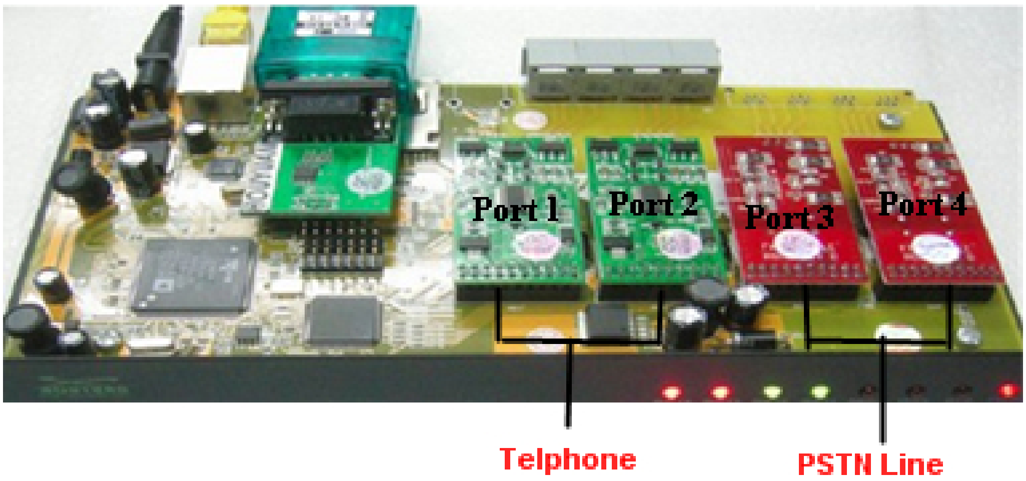
. If you forget

the IP Address of IP0x you have set up, you can use the fallback IP Address: 172.31.255.254/30. Before logging into IP0x, please set up the IP Address of your PC: 172.31.255.253 and SubMask: 255.255.255.252. At last, you can open your browser and enter:172.31.255.254 to log into the web page of IP0x.

2.4 Console Port Access to IP0x

If you do not have network connection between IP0x and PC, you can try to access to IP0x by console port. Please try to do as the following steps:

1. Please connect the console port of IP0x to your PC's console port with RS232 console cable, you can refer to the following illustration:



2. Please run your Hyper Terminal, and set up the console port like the following:
   Bits per second: 115200
   Data bits : 8
   Parity: None
   Stop bits: 1
   Flow control: None
3. Change the IP Address by Hyper Terminal

The default IP address of IP0x is 192.168.1.100. Your network may have a different IP address range such as 192.168.10.xx. In this situation, you can not access to IP0x by putty and browser if you do not change the IP0x IP address. So you have to change the IP address for IP0x by Hyper Terminal to make it in the same network segment as your LAN.

After you have accessed to IP0x by Hyper Terminal, please use the following command to change the IP address for IP0x.

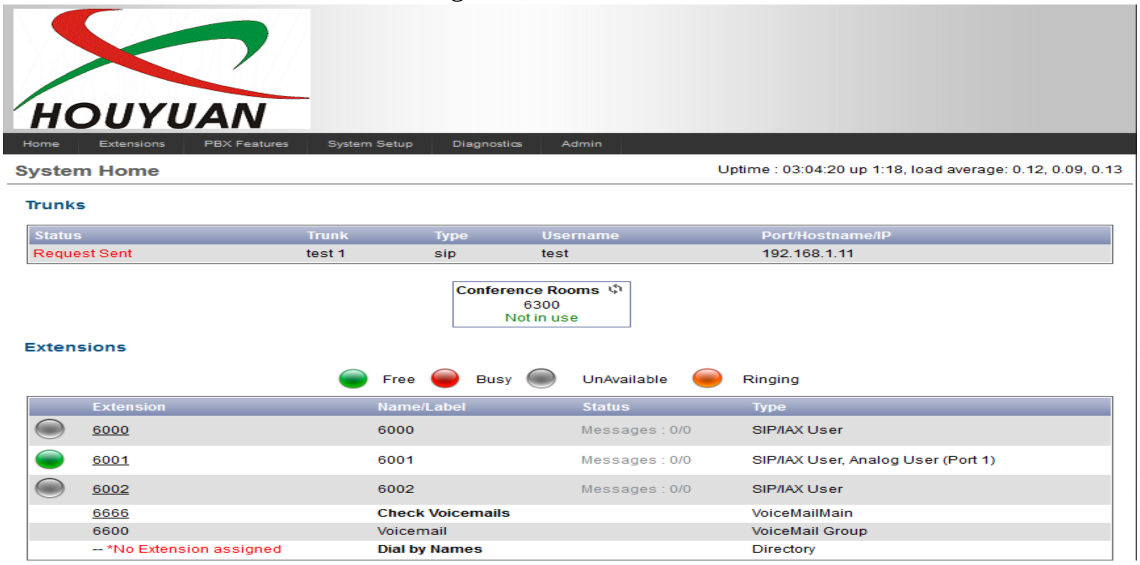
root:~> ifconfig eth0 192.168.1.151**(the IP address you want to set for IP0x)**

By this way, the IP address you set for IP0x is temporary, it will recover to the original default IP address after rebooting. If you want to give a static and permanent IP address for IP0x, you can try to set it in web GUI, for detail steps please refer to chapter 3.

Chapter 3 Configure IP0x by Web GUI

3.1 System Status

In the system status screen, it displays the functions you configured, such as: trunks, extensions, conference and so on like the following screen:



3.2 Configure Hardware

In the configure hardware page, it includes the following components: analog hardware, tone region, advanced settings.

Analog Hardware

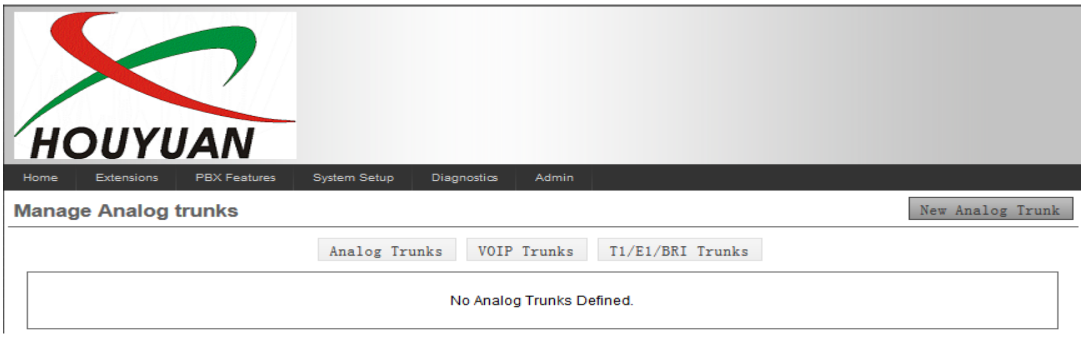
When you boot the IP0x, which will detect the FXO and FXS modules automatically, the analog hardware component displays the modules which are detected correctly.

Tone Region

You should select the tone region according to your country, if it does not have your country's name in the dropdown list, please ask your service operator which kind of tone region is used in your area.

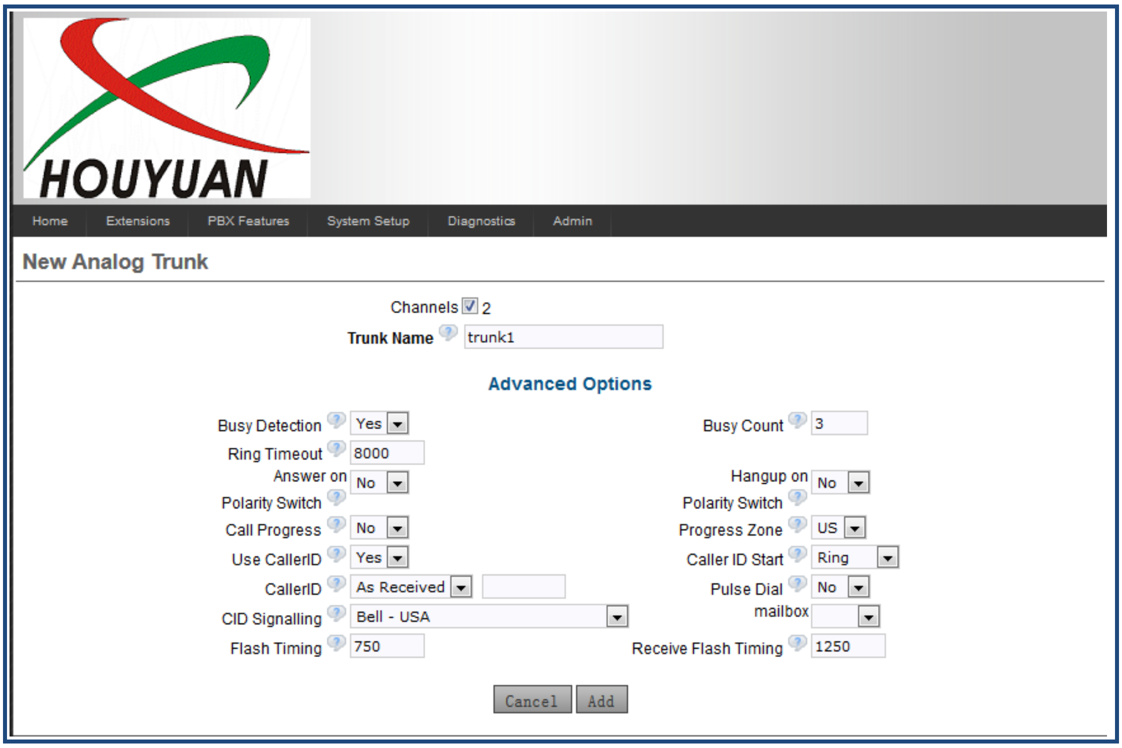
3.3 Trunks

To receive calls from PSTN and make calls to the outside world, you have to use trunk. Please select the **Trunks** option from the vertical menu on the left of the main page, then you can get the following screen:



3.3.1 Create Analog Trunks

Analog trunk is associated with FXO port, and it will call outside by PSTN line. Click on **New Analog Trunk** in the illustration above, the pop-up screen is where you create and set up trunk.



There are many parameters for you to set up, I just set the following two parameters:
Channels: select the FXO port you want to use. Here I use the port 2.
Trunk Name: a unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules. Here I use the trunk1 as my trunk name.

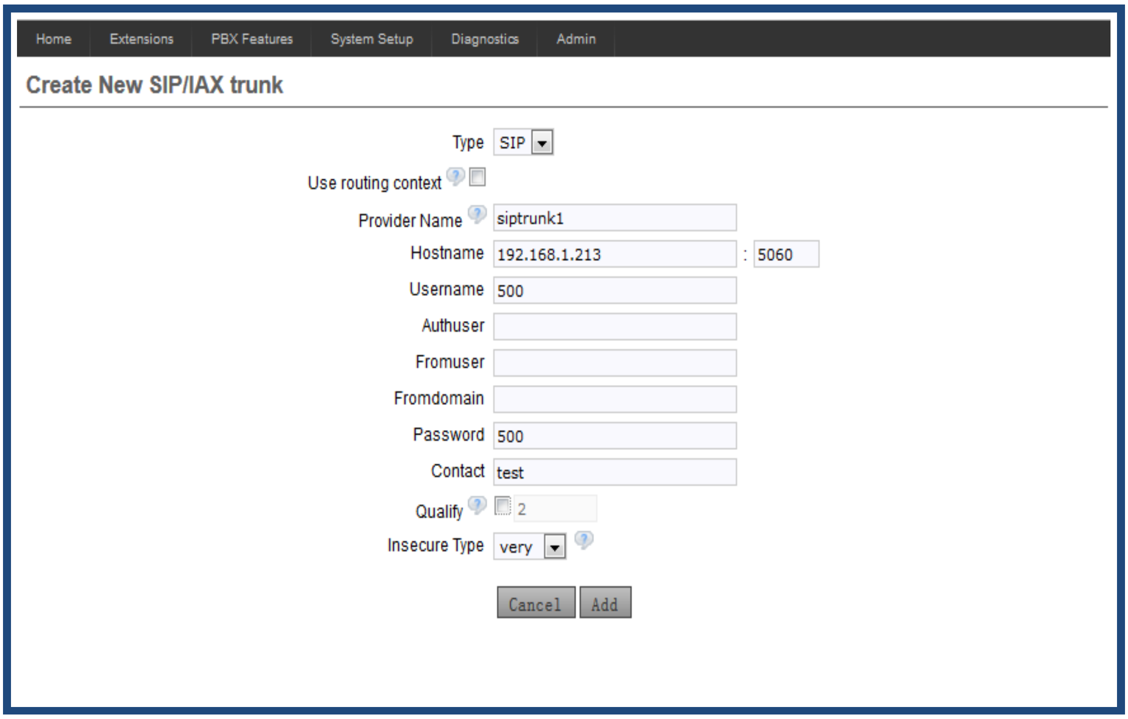
For the advanced options, you can put your cursor on the label, you can get the information of the parameter, customers have to set these parameters according to your service provider and your need.

3.3.2 VoIP Trunks

A VoIP service provider (VSP) that you have signed up with is also a trunk. Via the VoIP trunk you can dial via the VoIP service to reduce your cost when making international calls. You can set up

the VoIP trunk to make calls to the PSTN or other VoIP network depends on the service you use. You can also use the VoIP trunk to link headquarter and branch offices for free internal calls.

Click on **New SIP/IAX Trunk**, the following screen is where you create and set up VoIP trunk:



The important parameters are:

Type: You can select SIP or IAX type to meet your need.

Provide Name: a unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules.

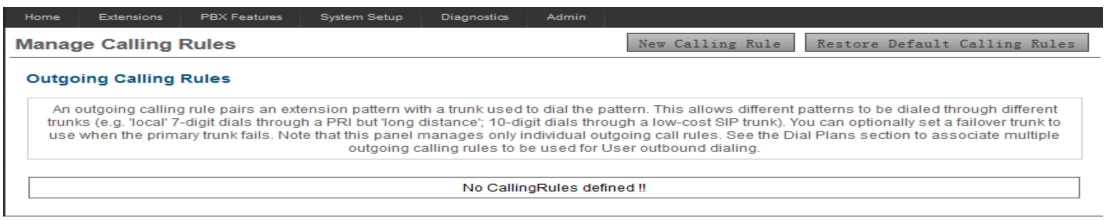
Hostname: the IP address or domain name of your service provider's server.

Username: the username that your service provider configured.

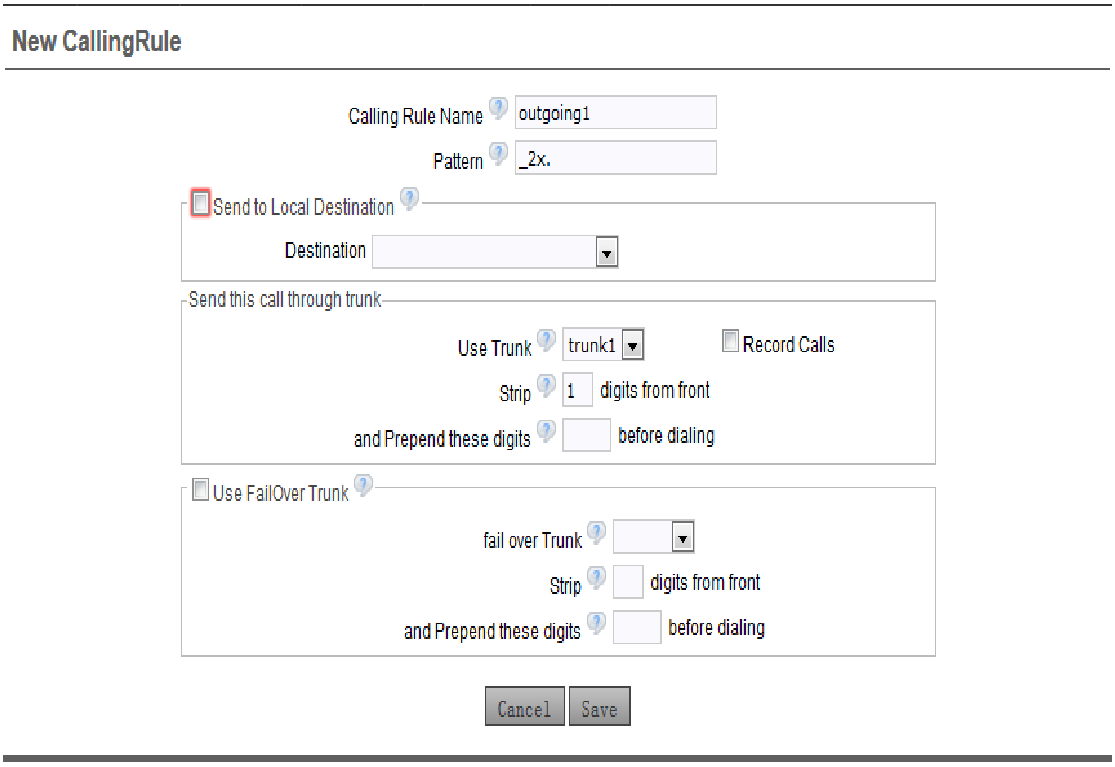
Password: the password that your service provider configured for the user.

3.4 Outgoing Calling Rules

Outgoing calling rules is used to route an outgoing call, when you make an external call, which trunk and what dial-pattern the call used are configured in outgoing calling rules. Please select the Outgoing Calling Rules option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Calling Rule** button on the illustration above, the following screen is where you create and set up outgoing calling rule:



The important parameters I configured are below:

Calling Rule Name: a unique label to help you identify the outgoing calling rule when listed in dial plans, I use outgoing1 as the calling rule name here.

Pattern: it acts like a filter for marching numbers you dialed, here I set up _2X., it means any number you dial out with prefix 2 will use this outgoing call rule.

Use Trunk: select the trunk for outgoing calling rule, here I select the trunk1 I set up before.

Strip: I press 1 here, it will strip the first digit of the number string you dialed.

You can get the detail information about every single parameter by putting your cursor on the  label.

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

The way of outgoing calling rules works:

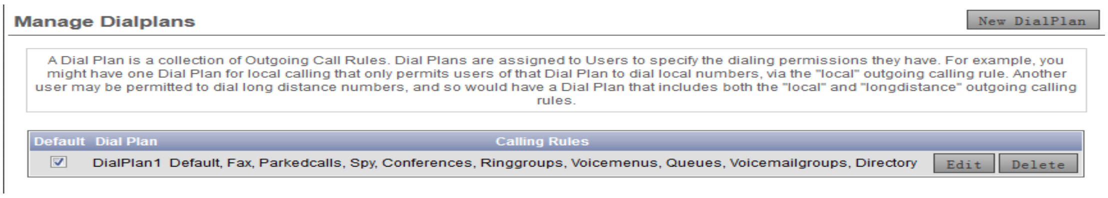
Every time you dial a number, asterisk will do the following in strict order:

- Examine the number you dialed.
- Compare the number with the pattern that you have defined in your first outgoing rule and if matches, it will initiate the call using that trunk. If it does not match, it will compare the number with the pattern that you have defined in the second outgoing rule and so on.
- Pass the number to the appropriate trunk to make the call.

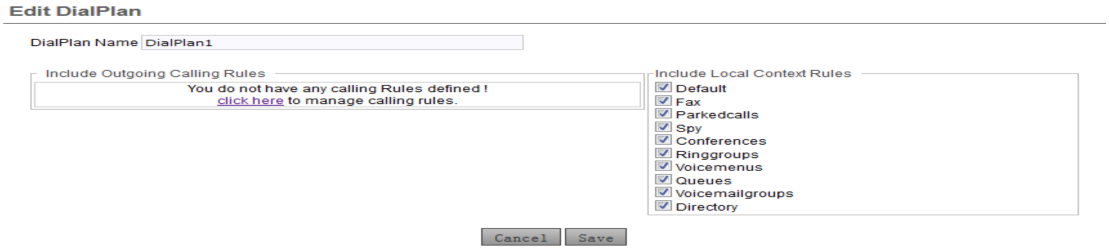
3.5 Dial Plans

A DialPlan is a set of Calling Rules that can be assigned to one or more users. Please select the

Dial Plans option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New DialPlan** button on the illustration above, the following screen is where you create and set up dial plan:



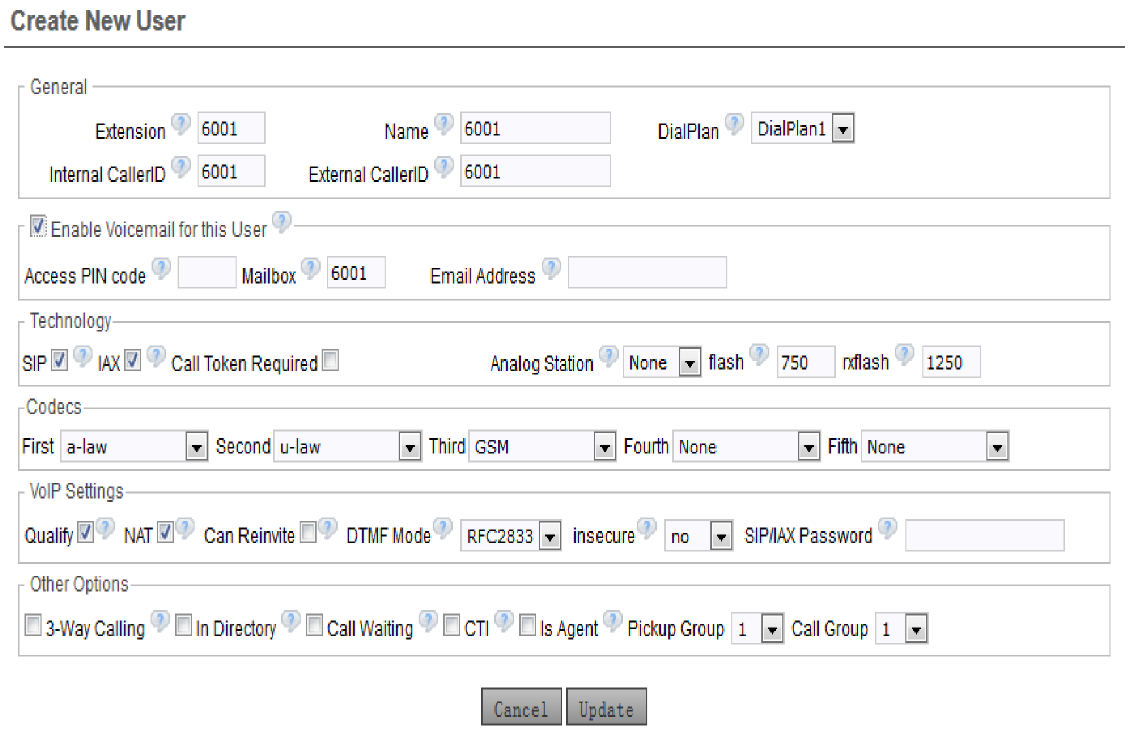
DialPlan Name: a unique label to help you identify the dial plan when listed in user component, you have to set up a dial plan name and select outgoing call rule and local context that you want to use.

3.6 Users

Users component is used to add or remove Analog, SIP, IAX extension. Please select the **Users** option from the vertical menu on the left, then you can get the following screen:

3.6.1 Create SIP/IAX User

Click on **Create New User** button on the illustration above, the following screen is where you create and set up user:



In General component, you have to set up Extension, CallerID, Name, OutBound CallerID parameters, and choose a DialPlan for the extensions. Here I set up user 6001, and select DialPlan1 for the user.

I select **Enable Voicemail for this User option**, so the user has voicemail function.

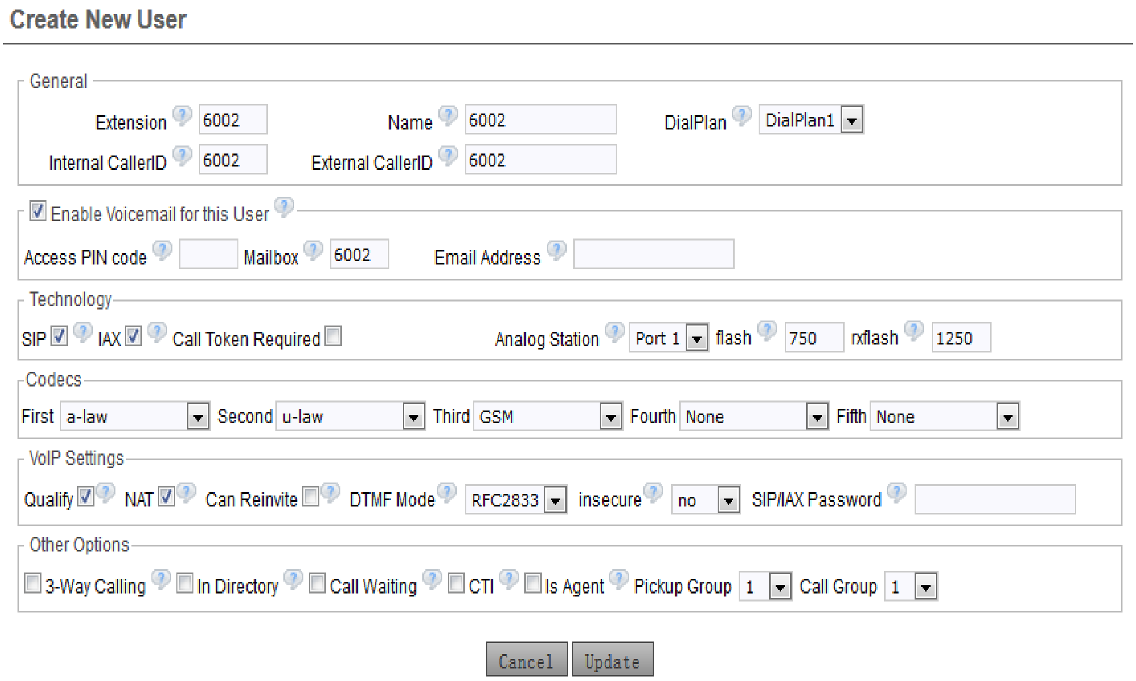
In the Technology component, you have to select SIP or IAX. Here I want to configure a SIP user, so I select SIP. For the Codec Preference, only the first two types of code you set are available.

In the **Other Options** component, I select **Is Agent** which will be listed in Call Queues as a selectable member for call queue.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.6.2 Create Analog User

Click on **Create New User** button, the following screen is where you create and set up user:



In the General component, you have to setup Extension, CallerID, Name, OutBound CallerID parameters, and choose a dialplan for the phone. Here I set up user 6002, and select DialPlan1 for the user.

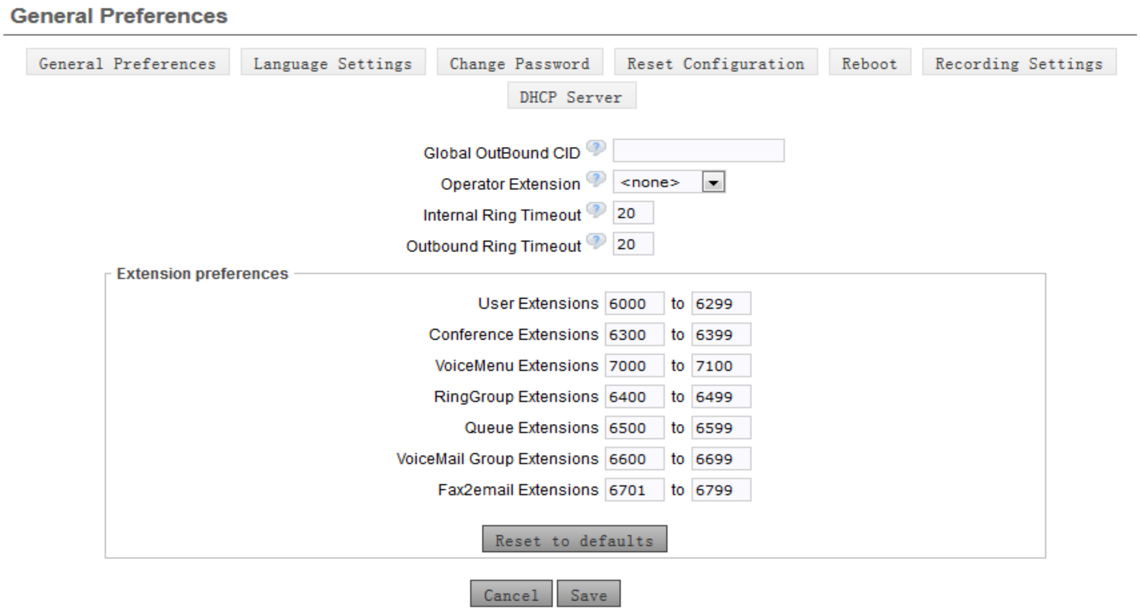
I select **Enable Voicemail for this User option**, so the user has voicemail function.

In the Technology componet, you have to select the port in which the analog phone will be plugged from the drop-down list of **Analog Station**. I select **Enable Voicemail for this User option**, so the user have voicemail function.

In the **Other Options** component, I select **Is Agent** which will be listed in Call Queues as a selectable member for call queue.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

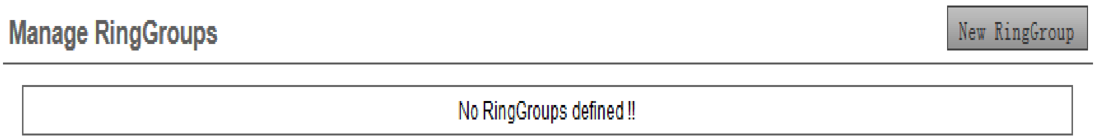
Attension: in the textbox of Extension, the value you set is limited to a range, you can adjust the range in the following screen to meet your requirement. Please select the **Options** option from the vertical menu on the left, then you can get the following screen:



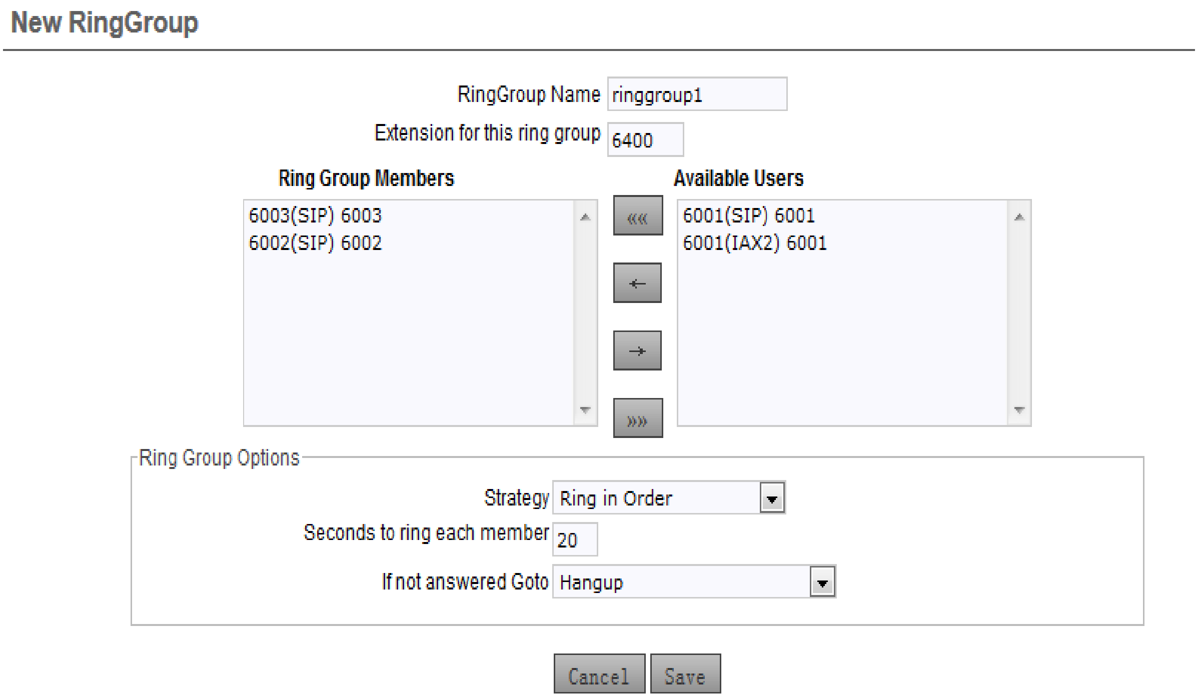
3.7 Ring Groups

Define Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Hunt groups.

Please select the **Ring Groups** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New RingGroup** button on the illustration above, the following screen is where you create and set up ring group:



Set the ring group name and extension for the ring group, select ring group members from available users.

Select strategy for ring group:

Ring in Order: when someone calls the ring group, the ring group member will ring in order.

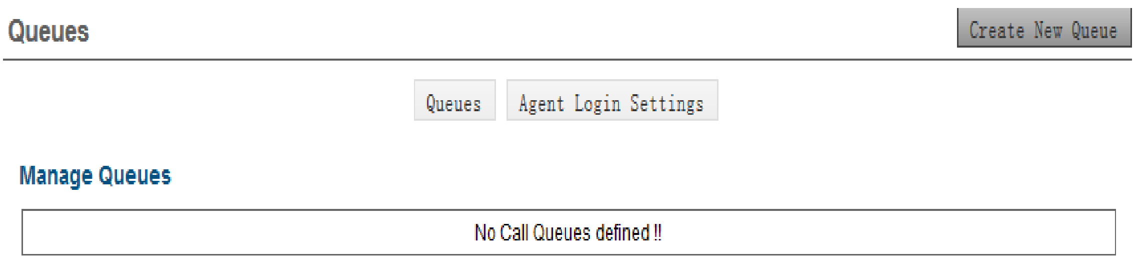
Ring all simultaneously: when someone calls the ring group, all of the ring group member will ring at the same time.

If not answered Goto: choose a destination from the drop-down list, when no one in the ring group answers the call.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

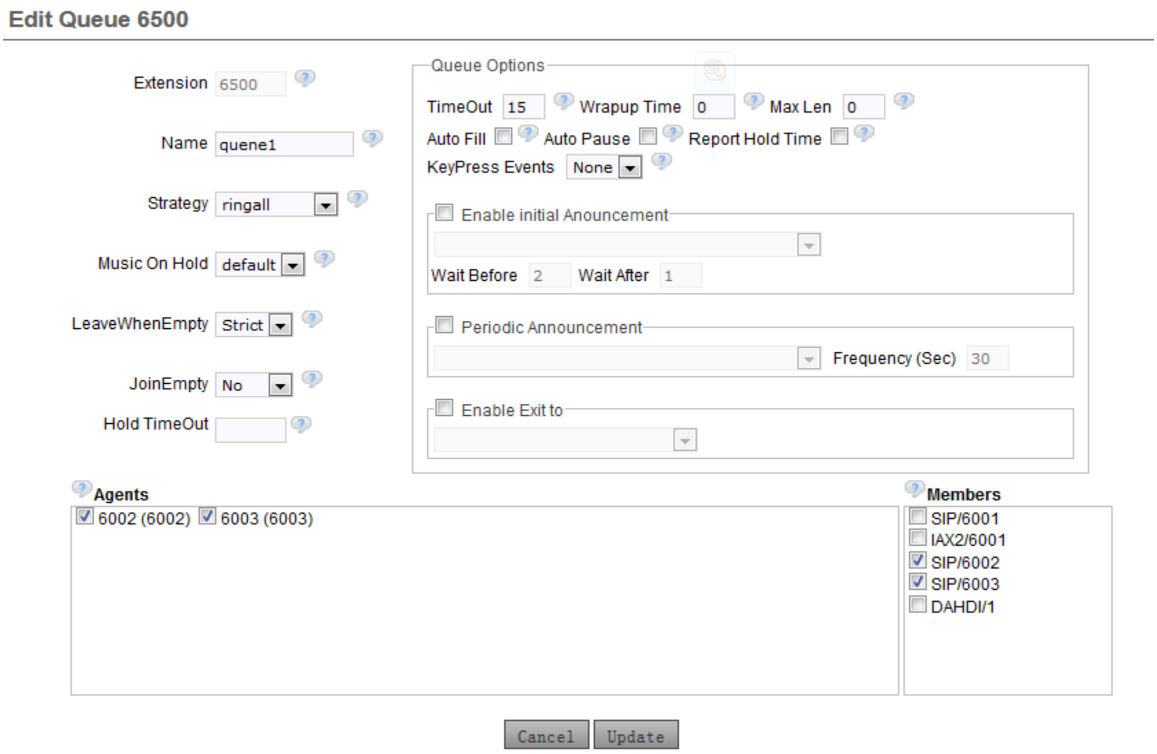
3.8 Call Queues

Please select the **Call Queues** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **Create New Queue** button on the illustration above, the following screen is where you

create and set up call queue:



Extension: a unique label to help you identify the call queue when listed in **outgoing calling rules** component.

Agents: select the users which you want them to be queue member.

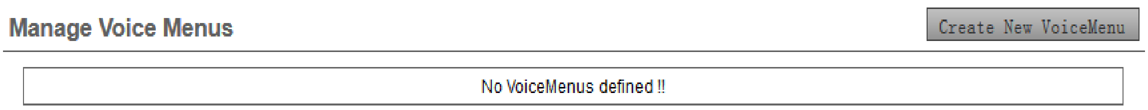
You can get information of other parameters by putting your mouse on the ⓘ label.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

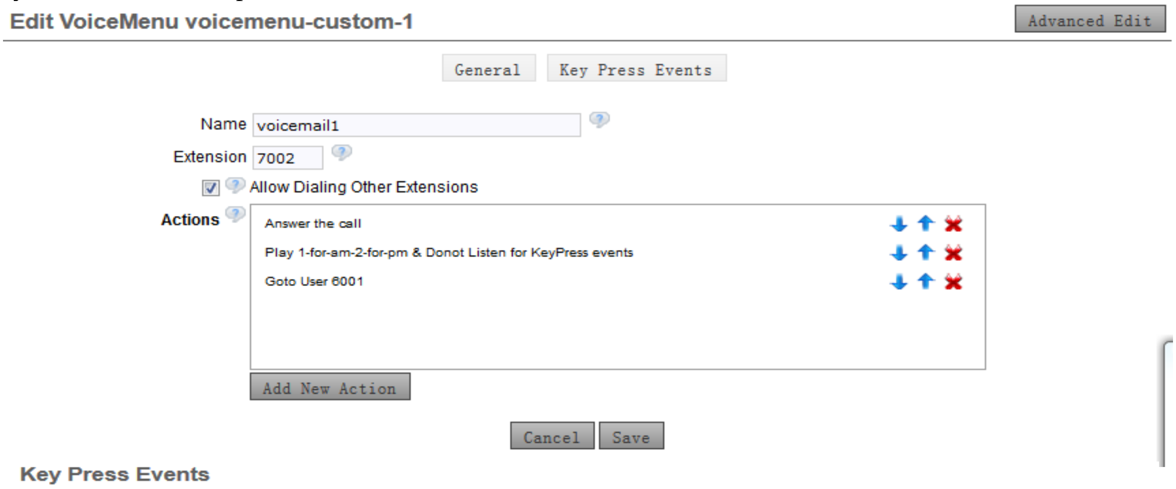
3.9 Voice Menus

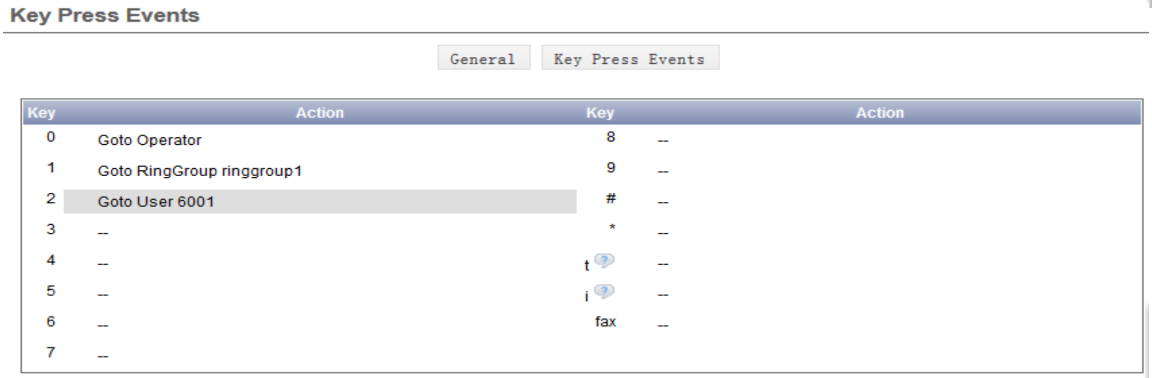
Like most organization, we would like to redirect all of the incoming calls automatically. The voice menu is very handy for these sorts of things. The system should allow callers to make the selection according to the voice menu.

Please select the **Voice Menus** option from the vertical menu on the left, then you can get the following screen:



Click on **Create New VoiceMenu** button on the illustration above, the following screen is where you create and set up voice menu:





Name: a unique label to help you identify the voice menu when listed in incoming calling rules.

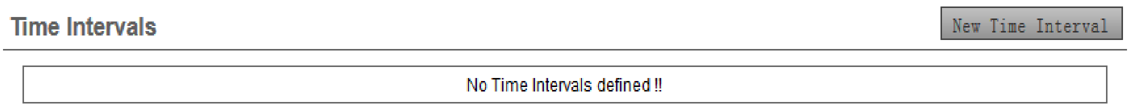
Add new Step: select an action from the drop-down list. I add three steps above, so it will answer the call, and play a sound file, at last go to user 6001.

Click on **Allow KeyPress Events**: when the caller is in voice menu, they can press some specific numbers which are defined here to enter other destination. Here I define three numbers for going to operator, ringgroup, and user respectively.

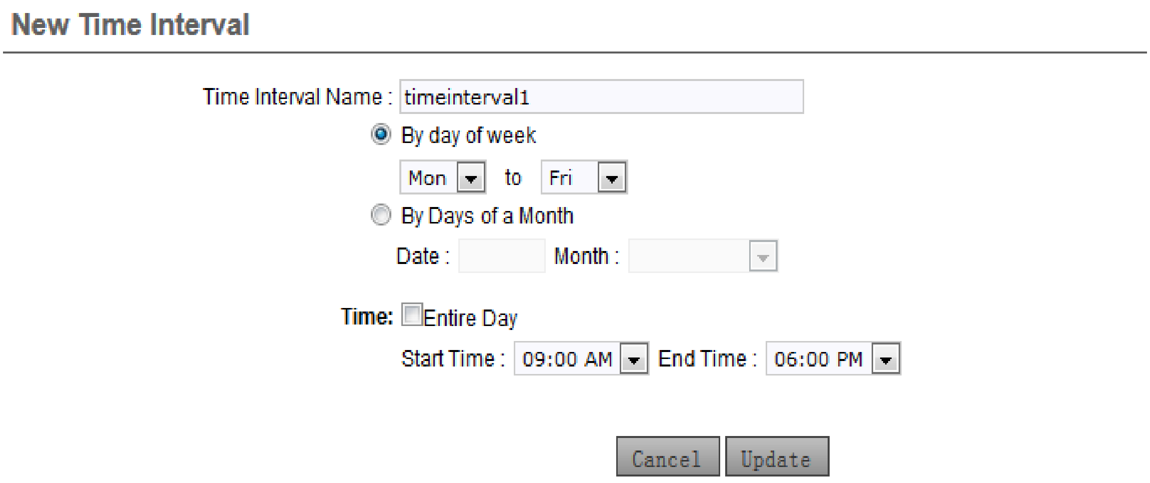
At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

3.10 Time Intervals

Time Intervals defines ranges of working time that will be used by call routing features. Please select the **Time Intervals** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Time Interval** button on the illustration above, the following screen is where you create and set up time interval:



Time Interval Name: a unique label to help you identify the time interval when listed in incoming calling rules. I set up timeinterval1 as time interval name.

By day of week: I select it from Monday to Friday, the incoming call rule only works from Monday to Friday.

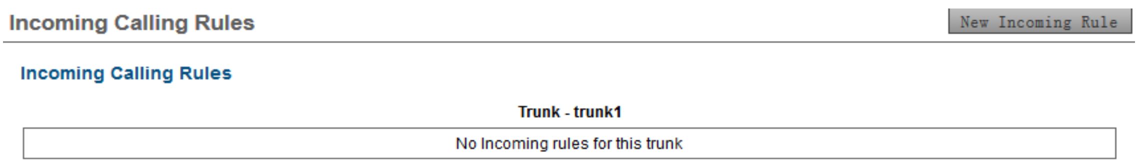
Time: I set up it from 09:00 AM to 06:30 PM, the incoming call rule only works from 09:00 AM to 06:30 PM.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

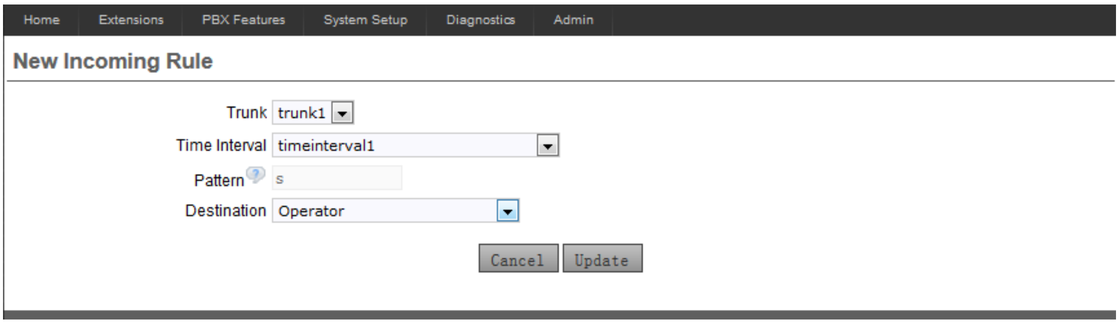
3.11 Incoming Calling Rules

This is where the behavior of incoming calls from all trunks is being handled. When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, digital receptionist, voice menu or queue. For this purpose, Incoming Calling Rules need to be set up.

Please select the **Incoming Calling Rules** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Incoming Rule** button on the illustration above, the following screen is where you create and set up time interval:



Trunk: select trunk for incoming call to use. I select trunk1 I set up before.

Time Interval: determine the time when the incoming call rule works, I select timeinterval1 I set up before.

Pattern: match the destination number, I use S which will match any destination number.

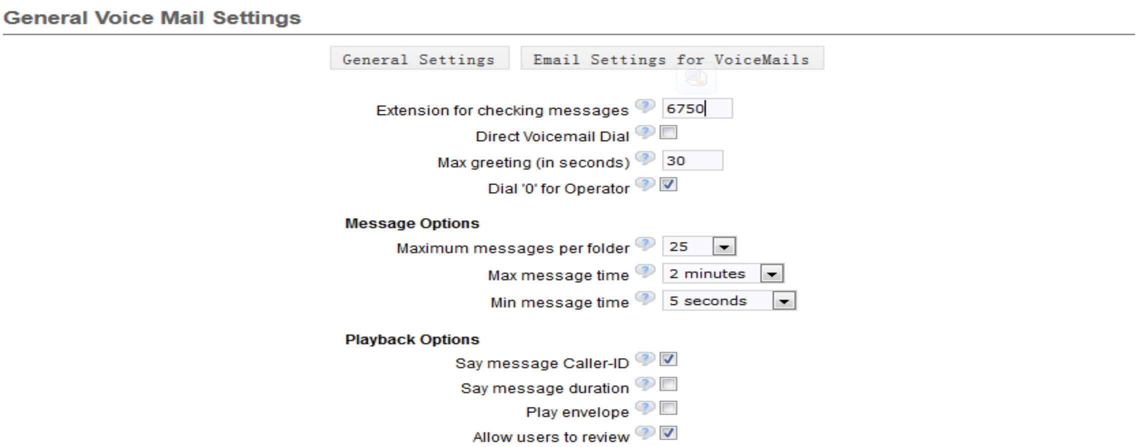
Destination: I select Operator, so the call will be ruled to Operator.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

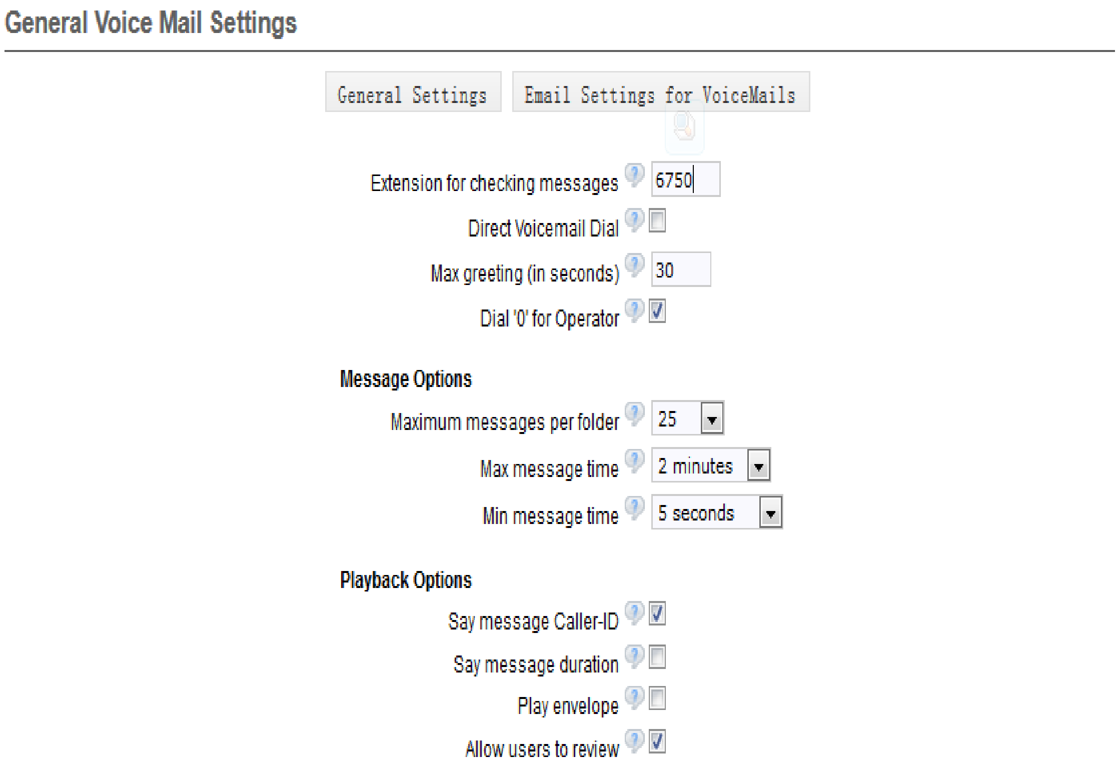
3.12 Voicemail

When you call someone who does not answer the call, you can leave a voice message for the called party if the called party supports voice mail.

Please select the **Voicemail** option from the vertical menu on the left of the main page, then you can get the following screen:

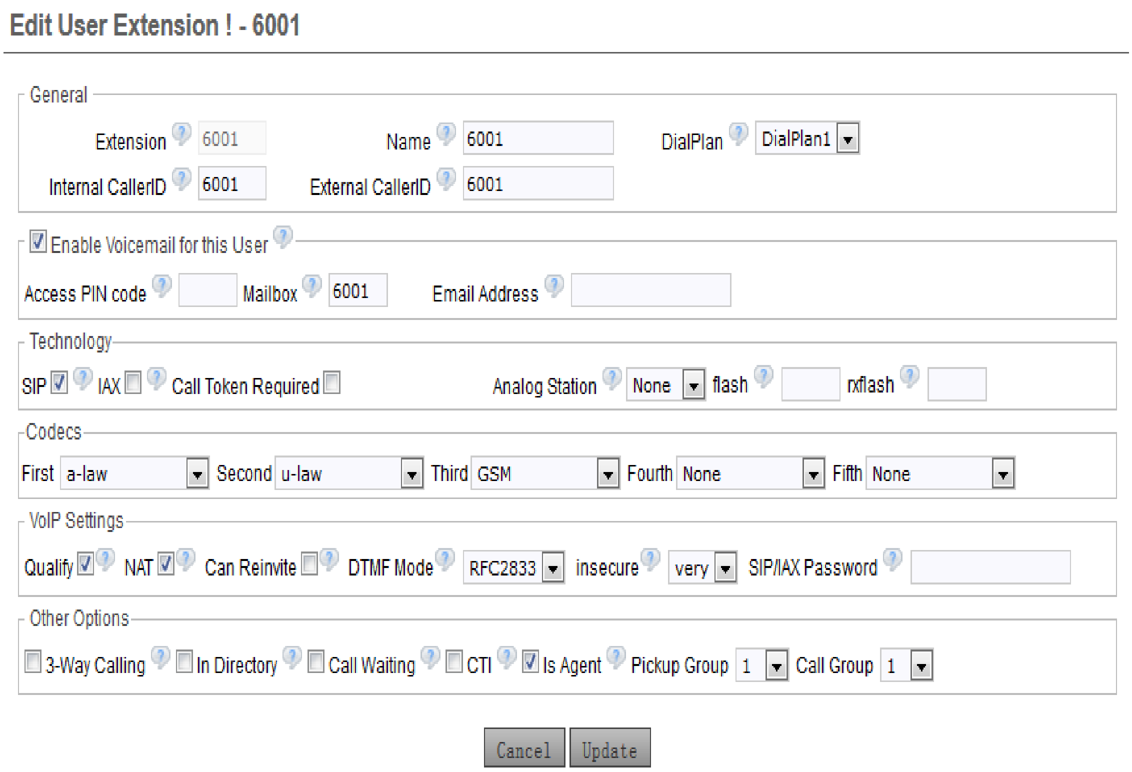


Click on **General Settings** button on the illustration above. You can see the following screen:



Extension for checking messages: when you dial 6750, you will hear the voice message other people left for you.

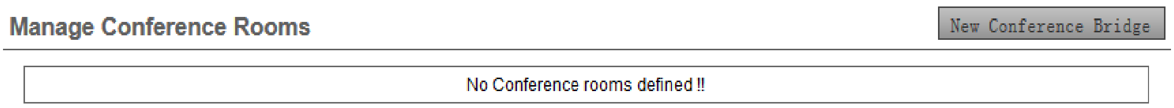
You can get information of parameters by putting your cursor on the label. If you want to set voicemail function for the user, you have to enable voicemail component when you set up a user. Please refer to the following illustration:



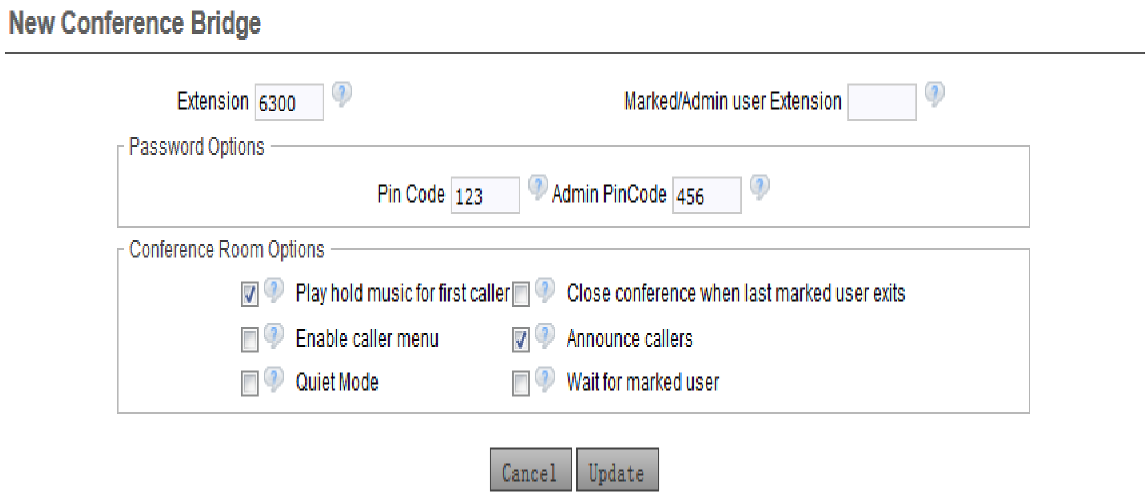
3.13 Conferencing

The conferencing function of Asterisk is similar to a Tele-conference call where multiple callers can call in and participate in a two-way conference like in a party room where everyone can talk and listen to one another or just to listen to a Tele-presentation.

Please select the **Conferencing** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Conference Bridge** button on the illustration above. Below is what my conference configuration page looks like:



Naturally there are some options that you may wish to have for the conference room. They are entirely up to you. The main important things are for you to create the conference room number and the conference pin code for you to know how to enter into the conference. The rest of the fields are optional. You can get information of other parameters by putting your mouse on the ⓘ label.

This conference number is 6300, the Pin Code is 123 for common member, the Pin Code is 456 for Admin. So you have to dial 6300 then, press the Pin Code, if you want to enter the conference. I enable the play hold music for option and announce callers option, so the first member who enter the conference will listen to a music and the online members will be informed when someone enter the conference.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

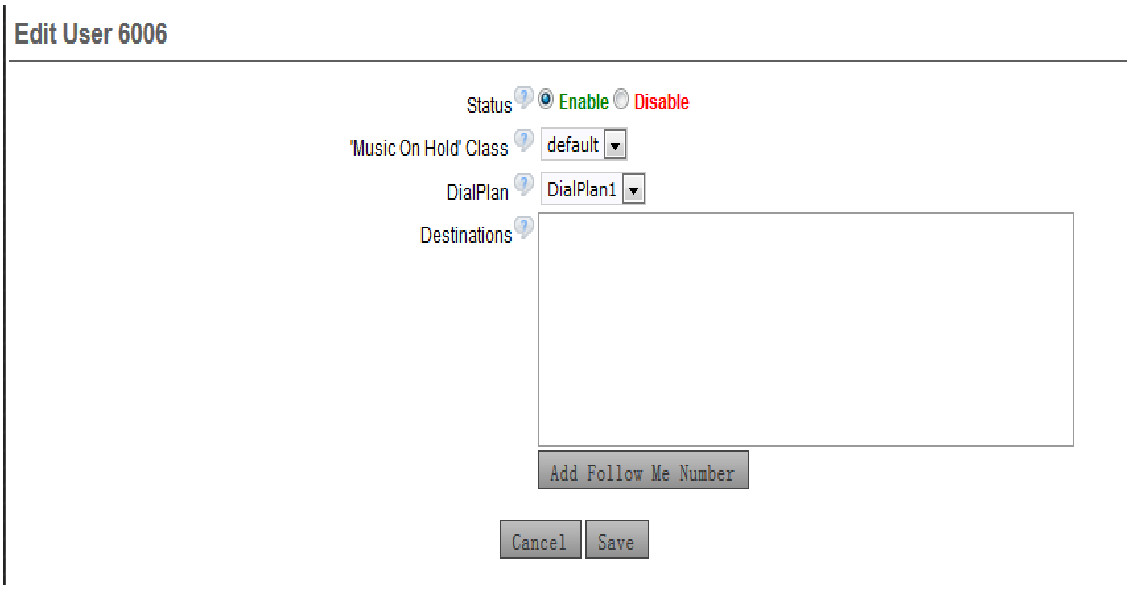
3.14 Follow Me

If A calls B, B does not answer, the call will be transferred to C who is set up in follow me.

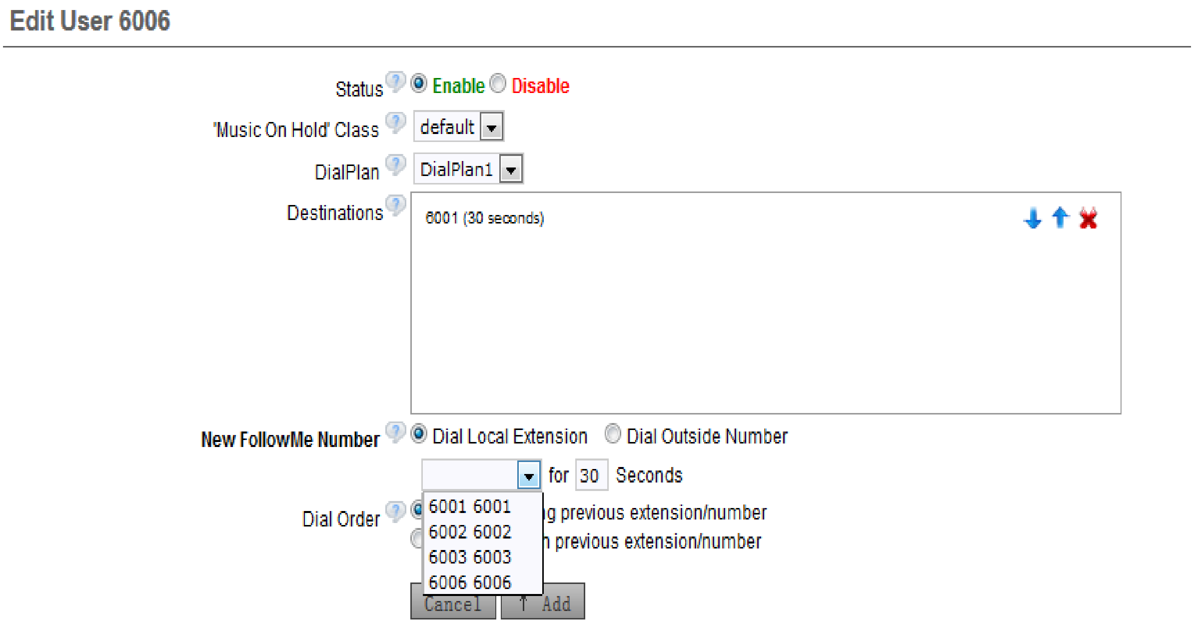
Please select the **Follow Me** option from the vertical menu on the left, then you can get the following screen:

You can choose user for which you want to setup follow me function, Here taking the user 6006

for an example, click on the **edit** button at the same line as 6006, you can get the following screen:



Select the **enable** status, and click on **Add FollowMe Number** button to add a destination phone.



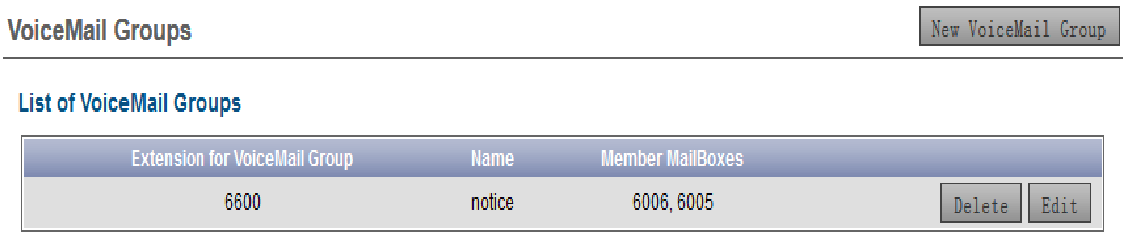
Click on **Dial Local Extension** and select 6001. Click on **Add** button and click on **Apply Changes** button in up right corner of the main page.

Through the above settings, someone calls 6006, but 6006 does not answer, the call will be transferred to 6001 automatically.

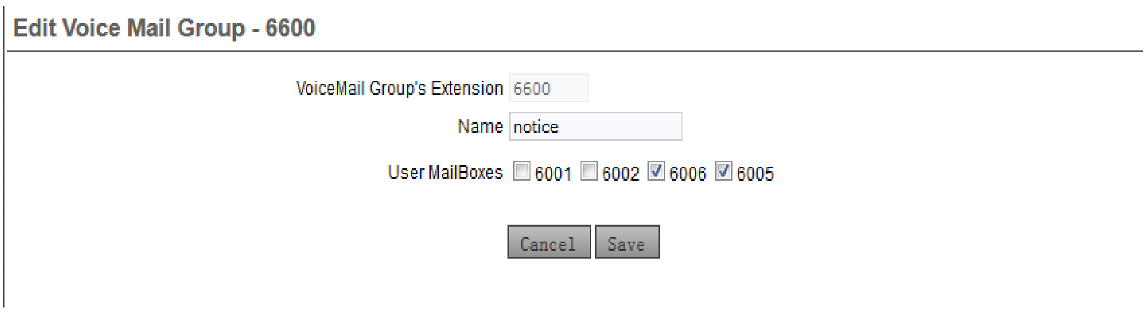
3.15 VoiceMail Groups

Define VoiceMail Groups to leave a voicemail message for a group of users by dialing an extension.

Please select the **VoiceMail Groups** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New VoiceMail Group** button on the illustration above. Below is what my VoiceMail Group configuration page looks like:

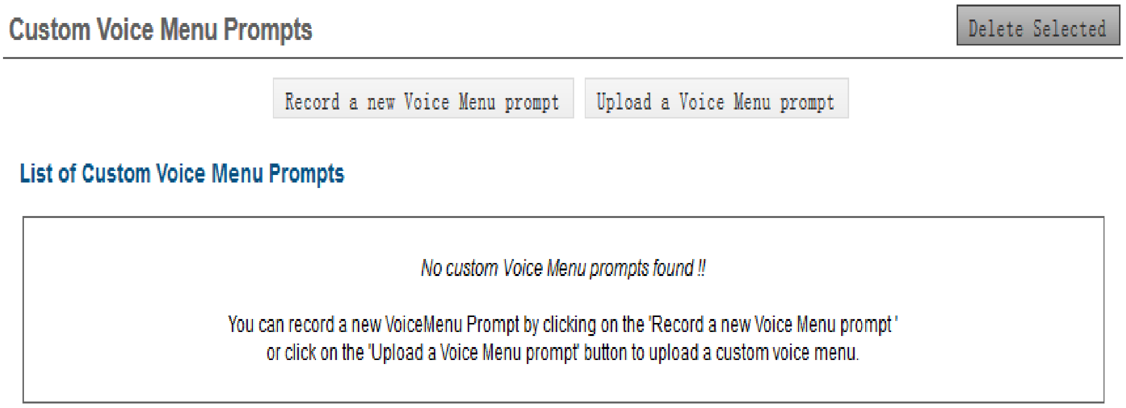


From the above settings, I can dial 6600 to leave message for user 6005 and 6006.

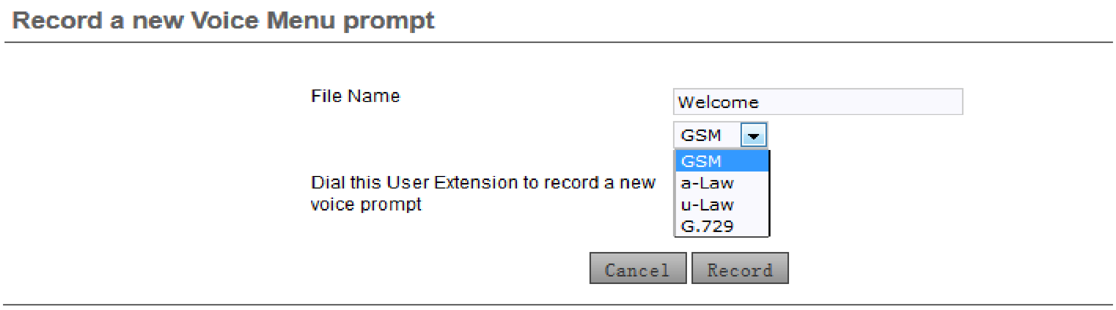
3.16 Voice Menu Prompts

This component is used for recording custom voice menu.

Please select the **Voice Menu Prompts** option from the vertical menu on the left of the main page, then you can get the following screen:



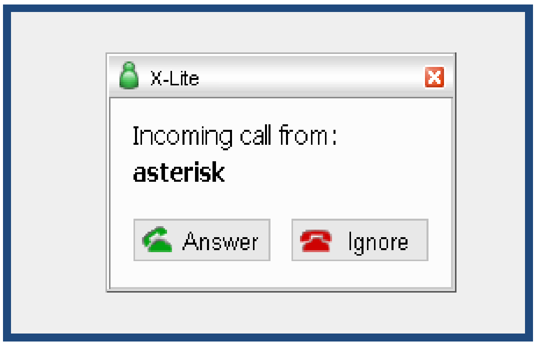
Click on **Record a new Voice Menu prompt** button on the illustration above. Below is what my Record a new Voice Menu prompt configuration page looks like:



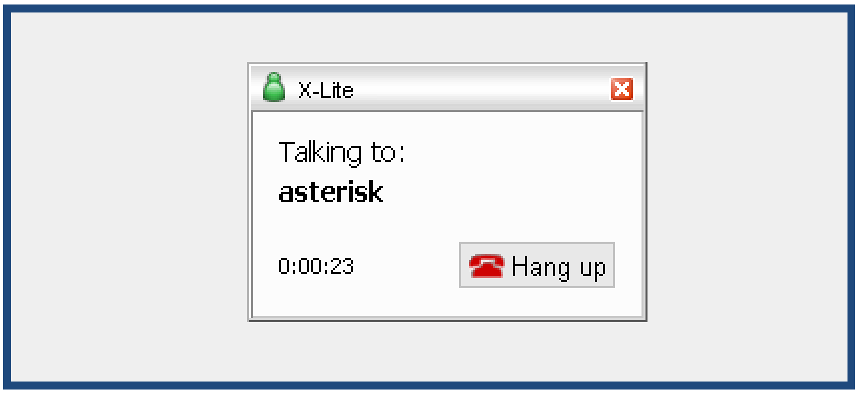
File Name: give a filename for the record sound file, here I give a name: Welcome

Dial this User Extension to record a new voice: dial to a user, then the user pick up the phone and speak the voice menu which will be recorded. Here I select 6001 I set up before.

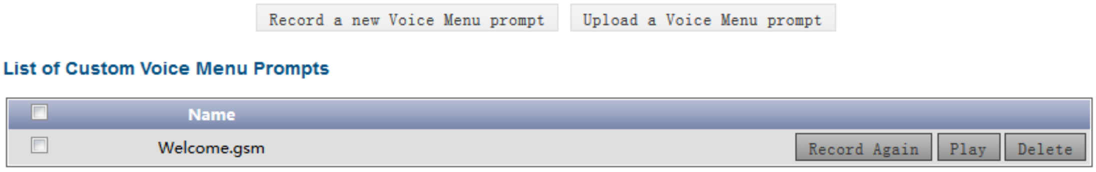
Click on **Record** button, the asterisk will call to 6001, 6001 will show like the following:



Click on **Answer** button, then you call speak and start to record what you say. The following illustration will be presented after you click on the **Answer** button.

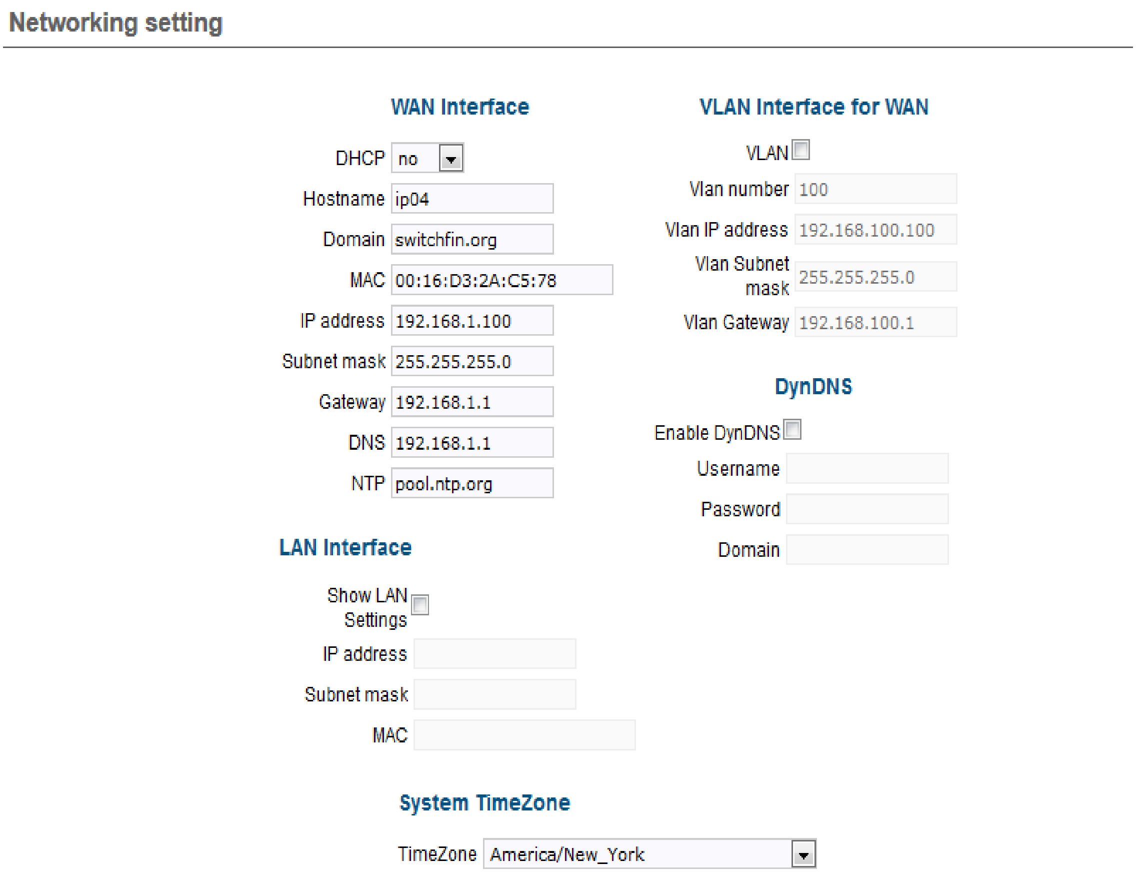


When you want to finish the record, please click on **Hang up** button.

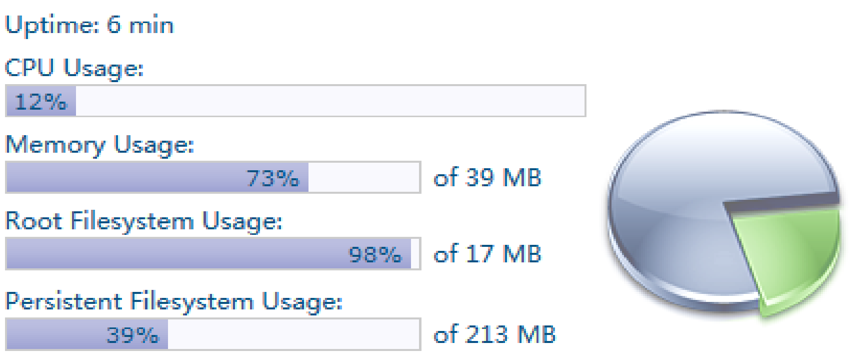


After you finish the recording, please refresh you webpage, and enter into **voice menu prompts** component again, you can see you have had a sound file like the above.

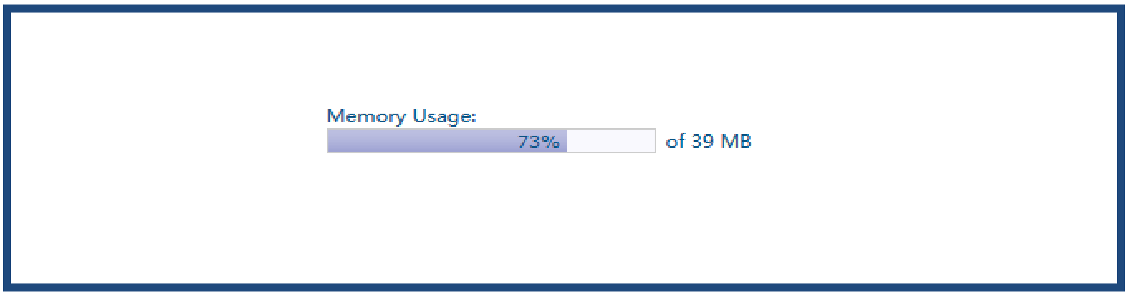
Network Information:



Disk Usage Information:

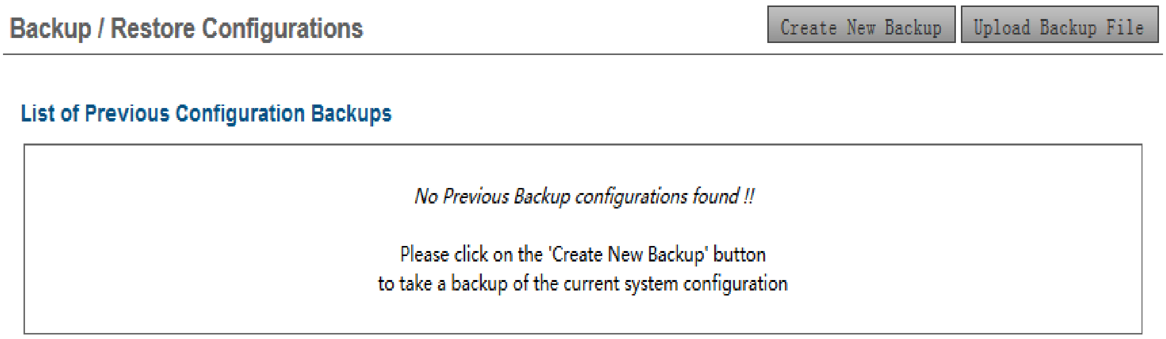


Memory Usage Information:

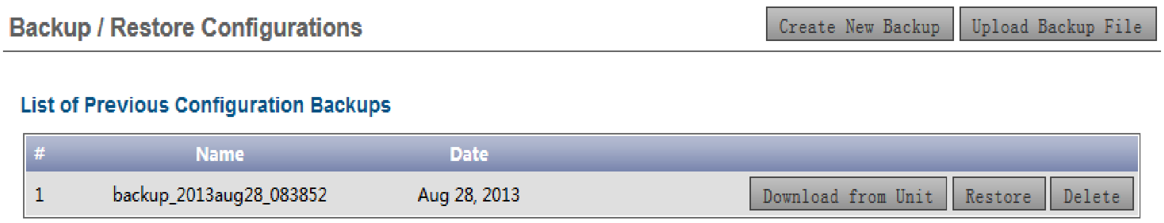


3.18 Backup

Backup and Restore are two of the mandatory functions of any application. IP0x is no exception. Customers can backup all the files under the /etc/asterisk/ directory and restore them.
Please select the **Backup** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **Create New Backup** button on the illustration above, you can get the following illustration:

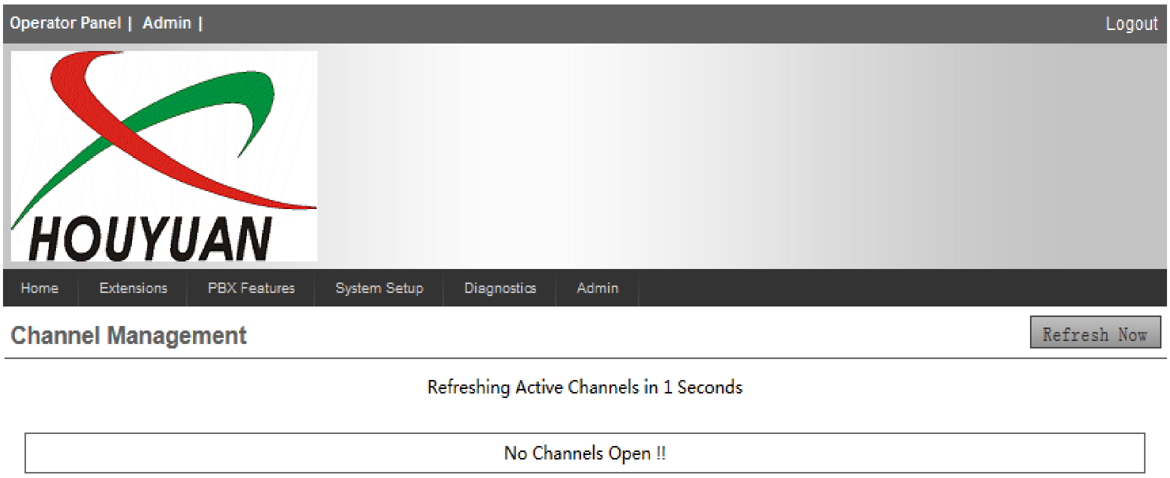


File Name: give a file name for the backed up file.
Click on **Backup** button, once the backup process is completed, you will see a screen with the backup filename displayed in illustration below.

Backup itself is not useful if it cannot be restored, IP0x also has this function. This is a very simple procedure. All you need to do is to click on the **Restore Previous Config** option.

3.19 Active Channels

The channels which are in communication status will be displayed in this component.
Please select the **Active Channels** option from the vertical menu on the left, then you can get the following screen:

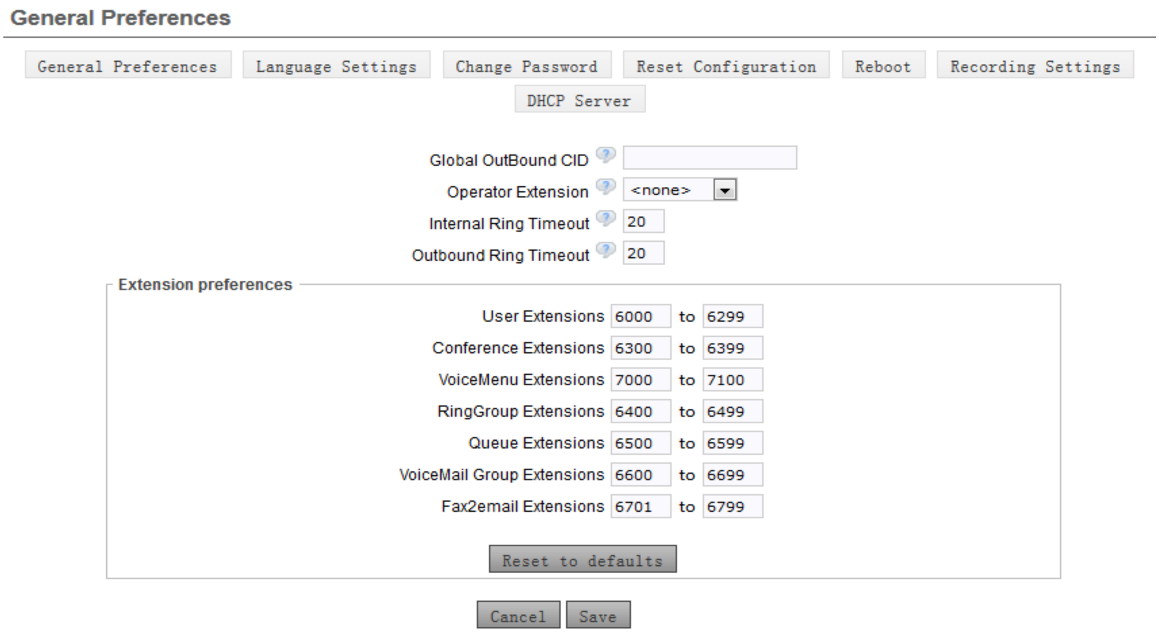


3.20 Options

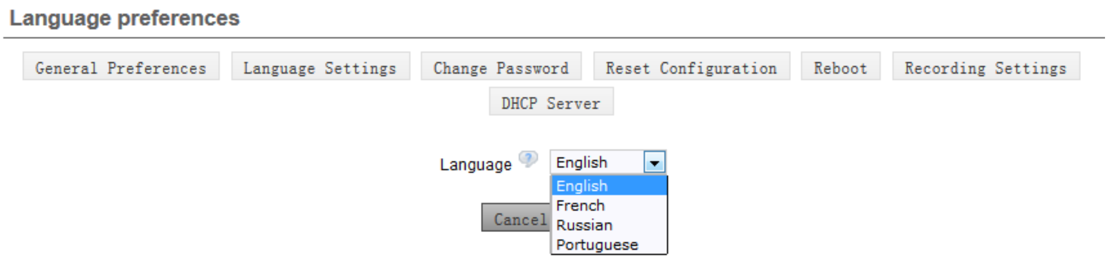
This component is used for administrator to manage the system, it includes the following modules:

General Preferences
Language
Change Password
Factory Reset
Reboot

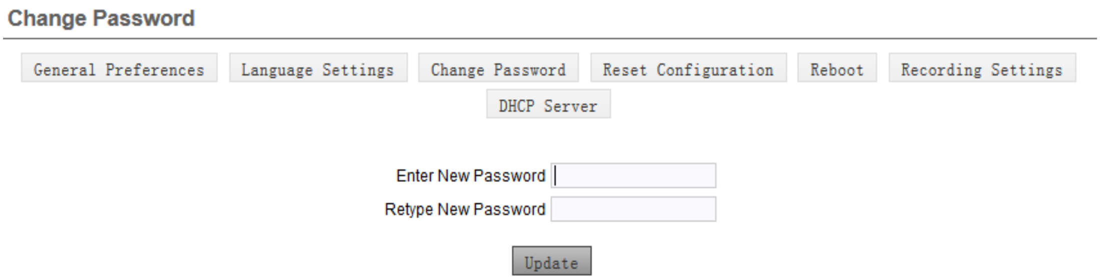
General Preferences: you can set up a user to be the operator and the range of extension number for different types' extensions like the following screen:



Language: change the sound file language in which they play.

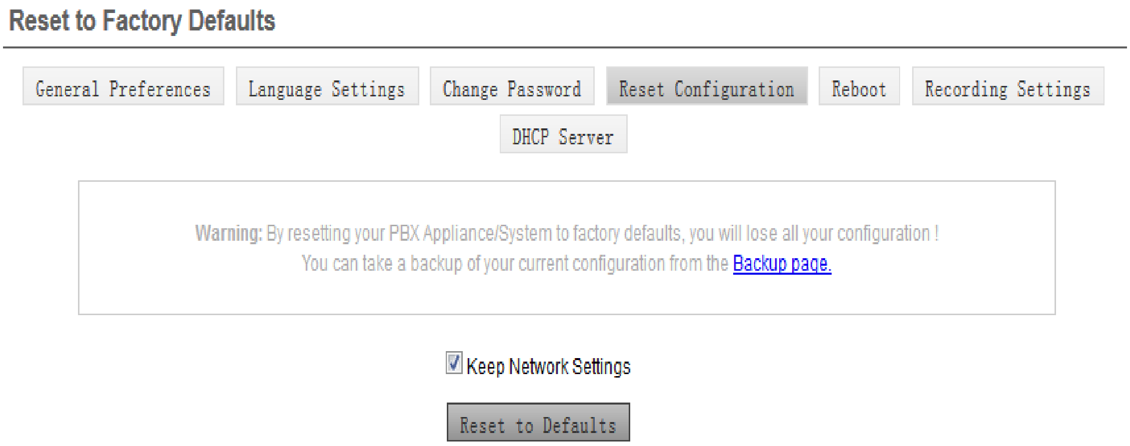


Change Password: it is used for customers to change the admin password, click on the **Change Password** button, the following illustration will be presented below:



After inputting your new password, please click on **Update** button, then click on **Apply Changes** button on the up right corner of the main page.

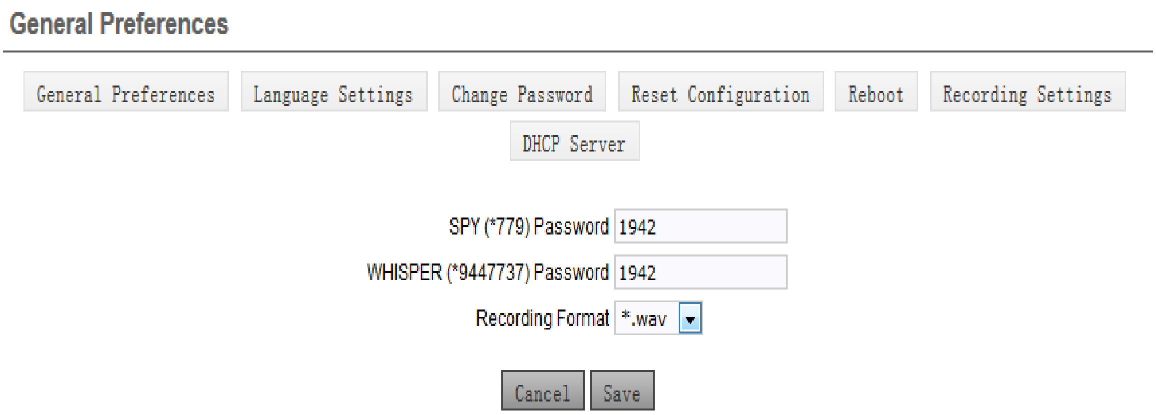
Factory Reset: it will help you to recover to the default factory settings. Click on **Factory Reset** button, the following illustration will be presented below:

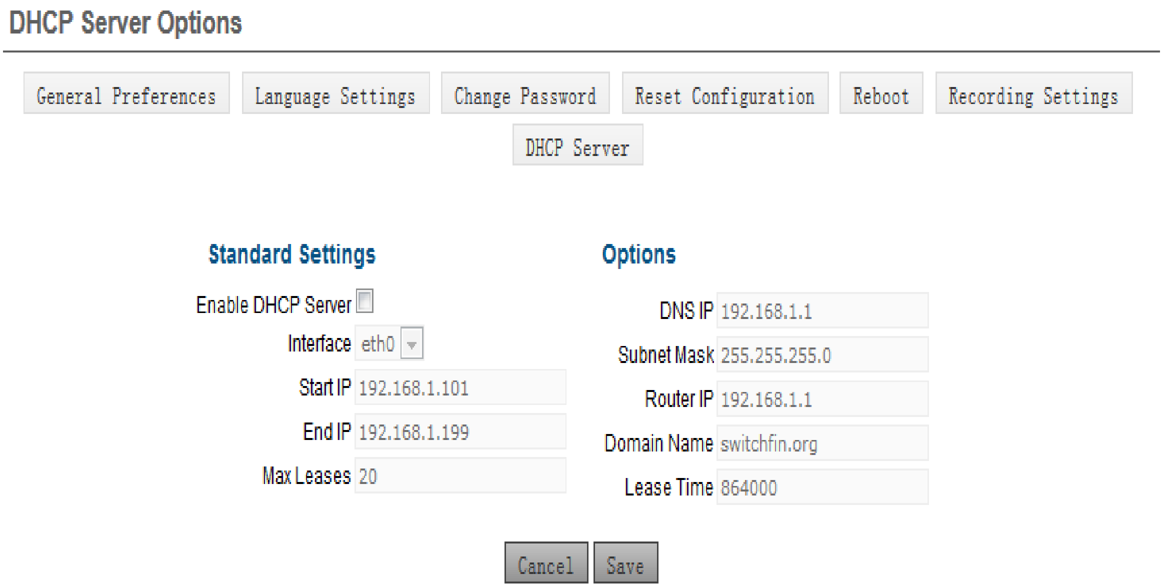


Please click on **Reset to Defaults** button to recover to default factory setting, then click on **Apply Changes** button on the up right corner of the main page.

Reboot: you can click on **Reboot** button→**Reboot Now** button to reboot your system.

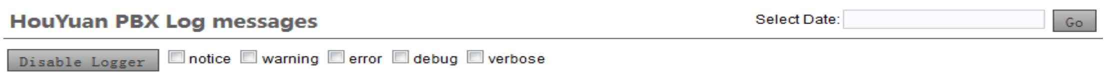
Recording Setting:



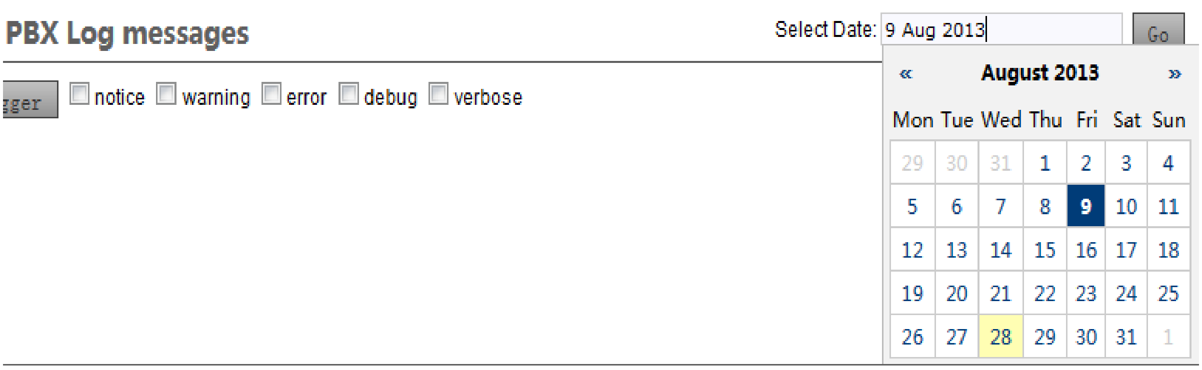


3.21 Asterisk Logs

After click on **Diagnostics> PBX Log messages** , please select the **PBX Logs** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on the textbox, you can get the following screen:



You can see a date table, and you can select the log to watch by clicking on the date. After choosing the date, please click on **Go** button, you can see the asterisk log of the day you choosed. Here I need to see the asterisk log of August 9st ,2013, I click on 9 in the date table, I get the following screen:

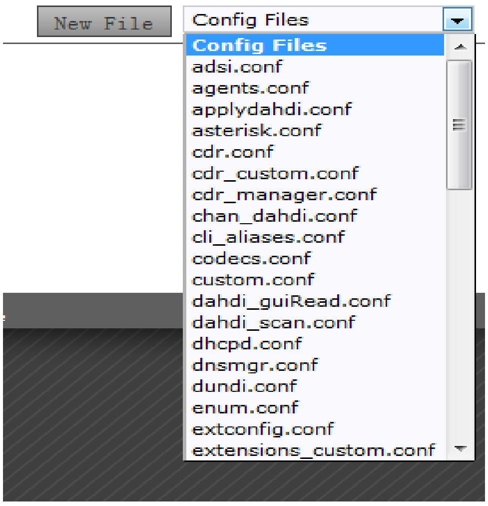
I click on **Go** button, then I get the log in the following screen:

```
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring insecure
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring signalling
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring macaddress
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring autoprov
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring label
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring linenumber
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring flash
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring disallow
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring allow
[Apr 21 03:45:16] WARNING[19680] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:45:36] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #1)
[Apr 21 03:45:40] WARNING[19691] ast_expr2.fl: ast_yyerror():  syntax error: syntax error, unexpected '=', expecting $end; Input:
[Apr 21 03:45:40] WARNING[19691] ast_expr2.fl: If you have questions, please refer to doc/channelvariables.txt in the asterisk source.
[Apr 21 03:46:06] WARNING[19691] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:46:26] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #2)
[Apr 21 03:47:16] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #3)
[Apr 21 03:47:46] WARNING[211] chan_sip.c: Maximum retries exceeded on transmission 24808208277904-20042119194381019 2.168.1.3 for seqno 1 (Critical Response
[Apr 21 03:47:46] WARNING[211] chan_sip.c: Hanging up call 24808208277904-200421191943810192.168.1.3 - no reply to our critical packet.
[Apr 21 03:48:06] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #4)
[Apr 21 03:48:56] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #5)
[Apr 21 03:49:46] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #6)
[Apr 21 03:50:36] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #7)
[Apr 21 03:51:26] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #8)
[Apr 21 03:52:16] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #9)
[Apr 21 03:53:06] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #10)
[Apr 21 03:53:56] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #11)
[Apr 21 03:54:46] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #12)
[Apr 21 03:55:36] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #13)
[Apr 21 03:56:26] NOTICE[211] chan_sip.c:    -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #14)
```

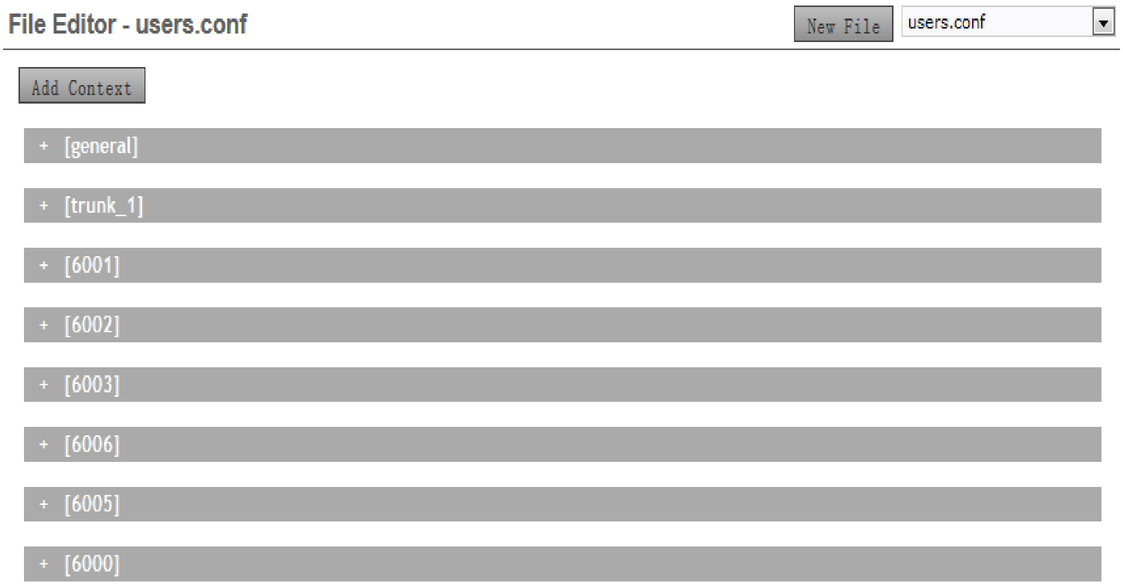
3.23 File Editor

please select the**File Editor** option from the vertical menu on the left, then you can get the following screen:

From the drop-down list of config files, you can select the file you want to edit or read.



Here I select users.conf file, so I can see the file and edit to meet my requirement.

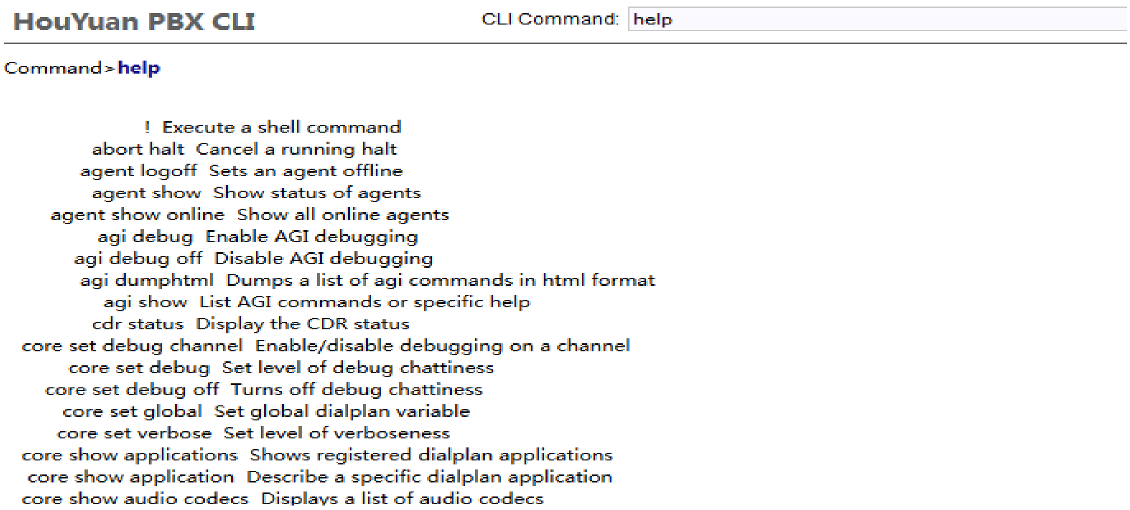


3.24 Asterisk CLI

These are some of the available CLI commands that can be executed from the console, you can input the asterisk CLI commands from the web page directly.

please select the

Asterisk CLI option from the vertical menu on the left, then you can get the following screen:



Here I input help command in the textbox, so I can get all the command which I can use in CLI mode.

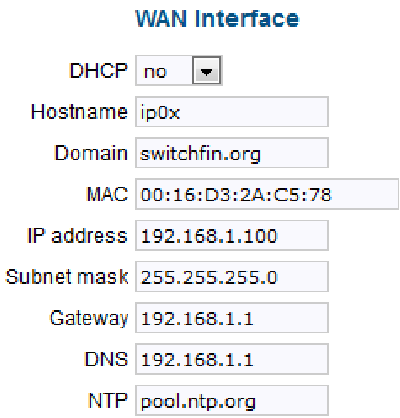
3.25 Network Settings

In order to give a static and permanent IP address for IP0x, you have to set it in web GUI. After you enter into the web GUI of IP0x, you can try to configure IP address according to the following

steps:

please select **Network Settings**

option from the vertical menu on the left of main page, the following screen is where you configure the network:



In the drop-down list of **DHCP**, you can see the following three options:

1. DHCP: yes: IP0x will obtain the dynamic IP address from your router.
2. DHCP: auto: IP0x will use the static IP specified below and ping the default gateway. When there is no response from the default gateway, the IP0x will switch to dynamically obtain the IP address from your router.
3. DHCP: no: IP0x will use the static IP address set below.

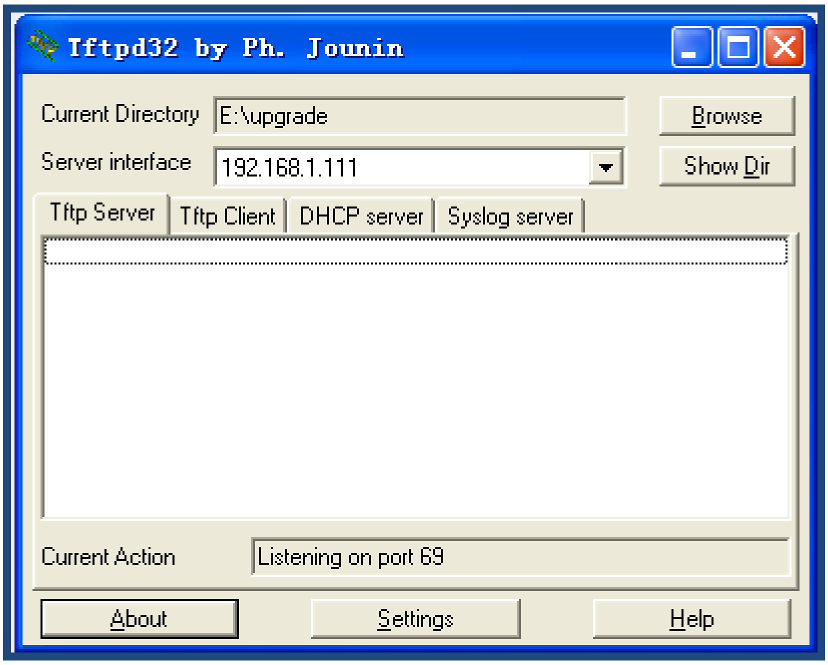
If you want to get static and permanent IP address, please do not select “yes”, after configure other parameters, please click “save” in the bottom of your page to save your setting.

3.26 Firmware Update

You can update to the latest version for IP0x by TFTP.

3.26.1 Download the Latest Firmware File and Set up TFTP Server.

1) Download the md5 file from http://www.houyuan.com/downloads/IPPBX/Firmware/IP02_08.md5, then put it in your TFTP server root directory.
2) Run your TFTP server, and I set up it like the following:

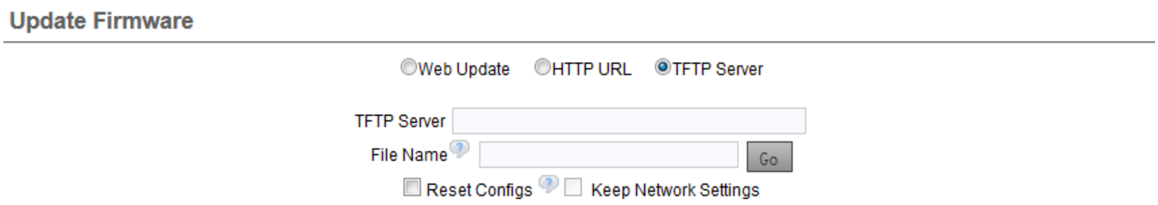


"E:\upgrade" is the root directory of my TFTP server, "192.168.1.111" is the IP Address of my TFTP server.

3.26.2 Update for IP02 from Web Page

please select **Firmware**

update option from the vertical menu on the left of main page, the following screen is where you update for IP02:

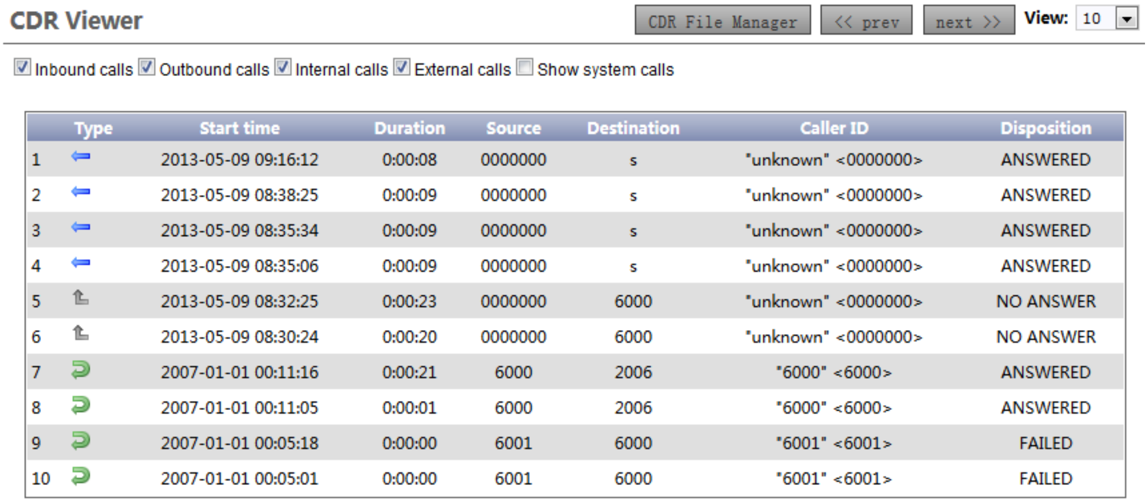


TFTP Server: enter the IP Address of your TFTP server in this textbox.
File Name: enter the update file name
Reset Configs: if you choose reset Configs, it will delete all of your configuration you have done before.
After setting up, please click on **Go** button to update for IP02.
Power off and power on the IP02, wait for several minutes. When the TEL port LED light up, it means the update is finished and you have the latest firmware.

3.27 Call Detail Records

This component provides the record of all incoming and outgoing calls including the channels used and duration of calls.

, please select the **Call Detail Records** option from the vertical menu on the left, then you can get the following screen:



CDR Viewer

CDR File Manager | << prev | next >> | View: 10

☑ Inbound calls ☑ Outbound calls ☑ Internal calls ☑ External calls ☐ Show system calls

	Type	Start time	Duration	Source	Destination	Caller ID	Disposition
1		2013-05-09 09:16:12	0:00:08	0000000	s	"unknown" <0000000>	ANSWERED
2		2013-05-09 08:38:25	0:00:09	0000000	s	"unknown" <0000000>	ANSWERED
3		2013-05-09 08:35:34	0:00:09	0000000	s	"unknown" <0000000>	ANSWERED
4		2013-05-09 08:35:06	0:00:09	0000000	s	"unknown" <0000000>	ANSWERED
5		2013-05-09 08:32:25	0:00:23	0000000	6000	"unknown" <0000000>	NO ANSWER
6		2013-05-09 08:30:24	0:00:20	0000000	6000	"unknown" <0000000>	NO ANSWER
7		2007-01-01 00:11:16	0:00:21	6000	2006	"6000" <6000>	ANSWERED
8		2007-01-01 00:11:05	0:00:01	6000	2006	"6000" <6000>	ANSWERED
9		2007-01-01 00:05:18	0:00:00	6001	6000	"6001" <6001>	FAILED
10		2007-01-01 00:05:01	0:00:00	6001	6000	"6001" <6001>	FAILED

You can click on the **prev** to look up the last page for call record, and click on the **next** to look up the **next** page for call record, you can also set the value from the drop-down list of **view** which means how many calls will be displayed in one page.

Chapter 4 an Application Case of IP PBX02

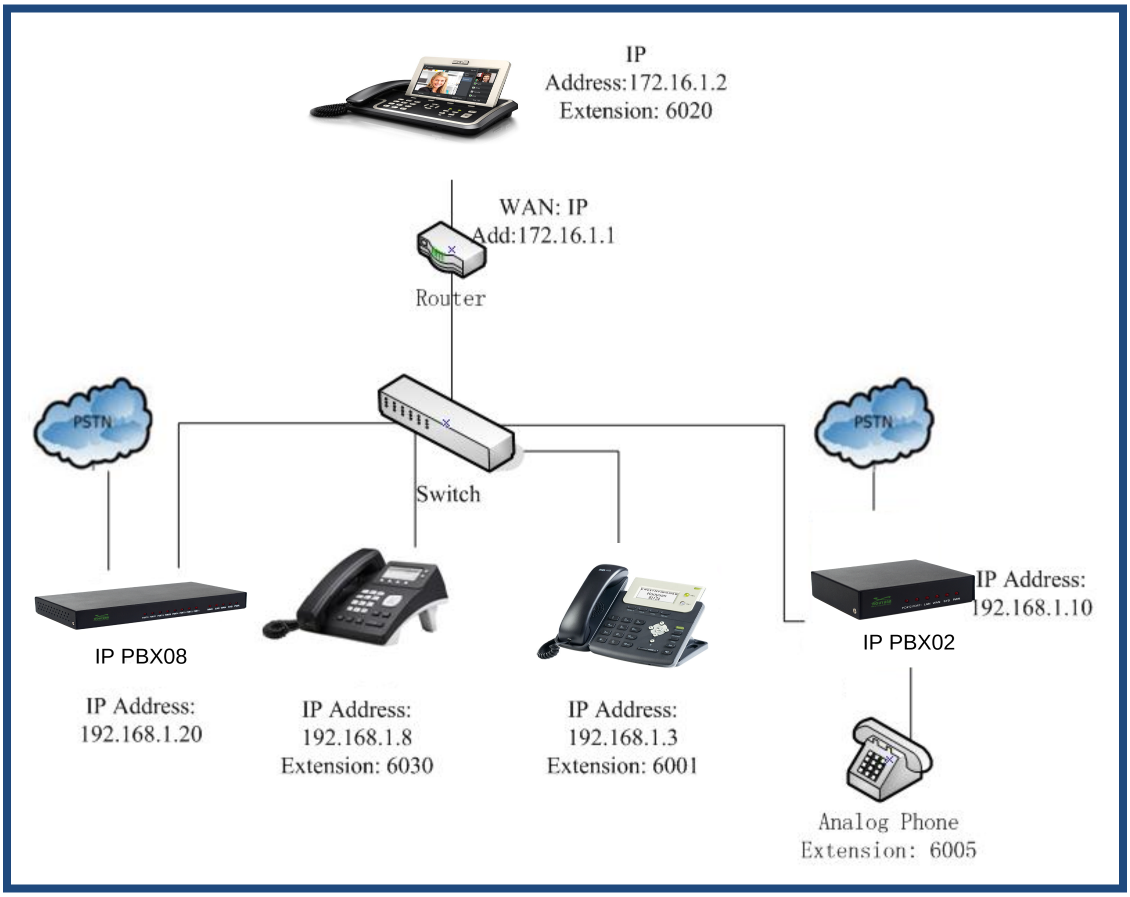


Figure: Network Topology

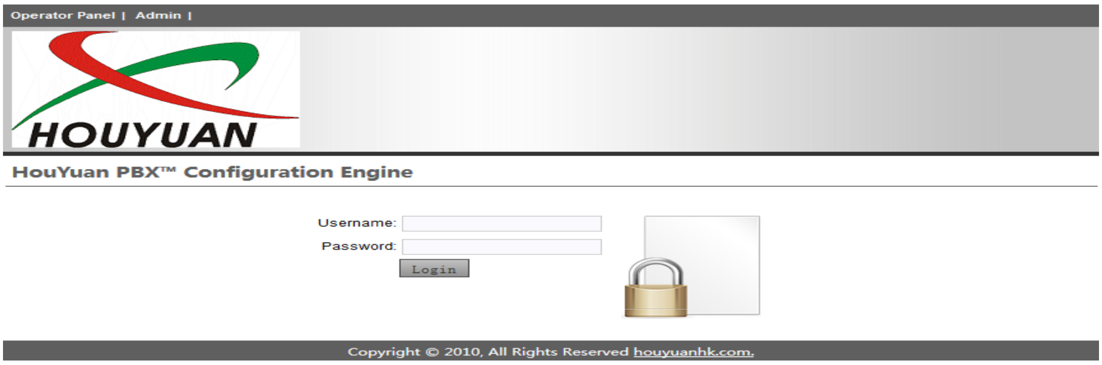
In the network topology above: user 6020 and user 6001 will be registered to IP02, user 6030 will be registered to IP08, analog phone 6005 is connected to FXS port of IP02. After configuration, it will realize the following function:

1) The internal user 6005 and user 6001 can call each other directly.
2) 6005 and 6001 can dial-out through IP02 to PSTN.
3) 6005 and 6001 can get incoming calls from PSTN by IP02.
4) 6030 can call-out to PSTN and get incoming call from PSTN through IP08.
5) User 6001 and 6030 can call each other through VoIP trunk, although they are registered to different IP PBX.
6) User 6020, 6005 and 6001 can call each other directly, although they are not in the same network segment.

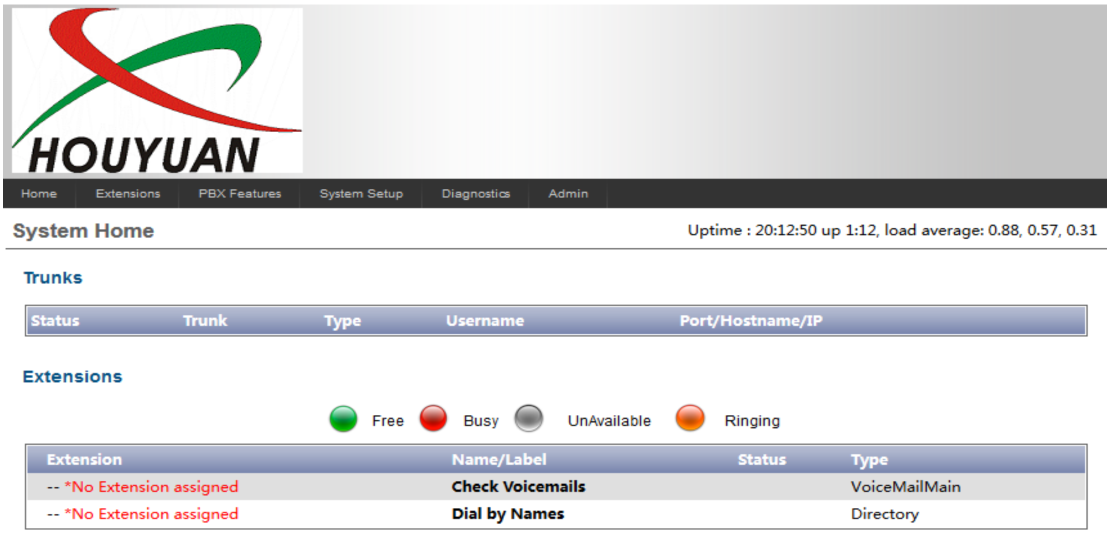
4.1 How to Make Internal Calls through IP0x

4.1.1 Access to the Web Page of IP0x by Browser

After connecting IP0x to LAN, please open your browser of PC with windows OS and input the IP Address of IP0x (the default IP address is 192.168.1.100), then you can get the following screen:



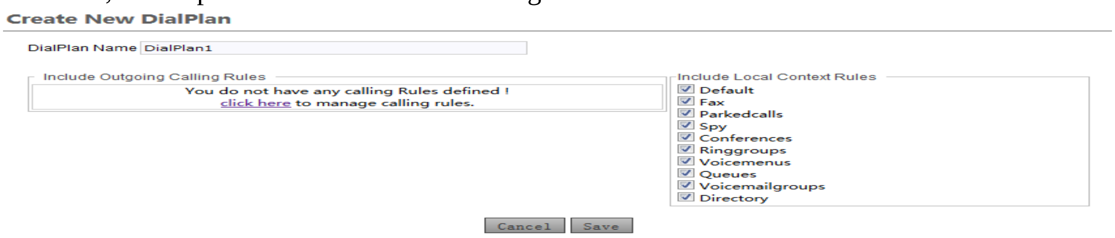
Please input the default Username: admin; Password: admin in the presented screen above. When you login successfully, you can get the configuration web page as below:



4.1.2 Add up Users from Web Page of IP0x

1) Add up a DialPlan

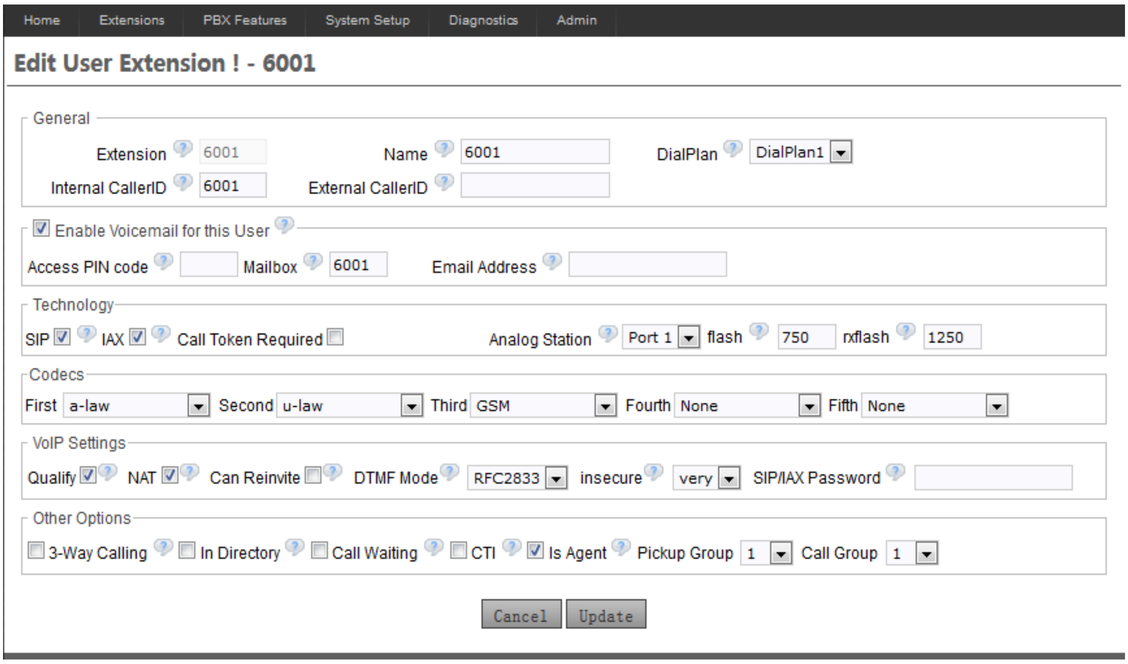
Before you add up user, you have to add up a DialPlan, please click on **Dial Plans→New DialPlan**, I add up a DialPlan like the following:



After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

2) Add up SIP user 6001

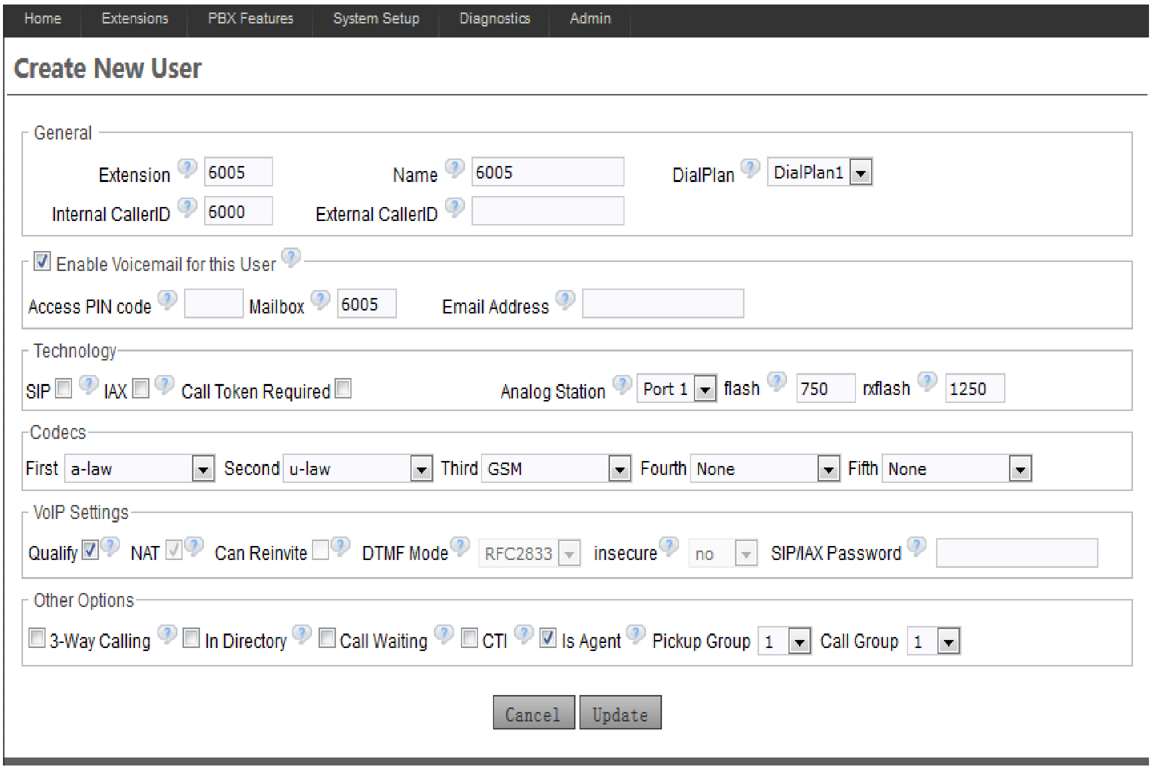
After logging into the web page of IP0x, please click on **Exten→ Create New User**, I configure user 6001 like the following:



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3) Add up an Analog user 6005

After logging into the web page of IP0x, please click on **Exten→ Create New User**, I add a user 6005 like the following:

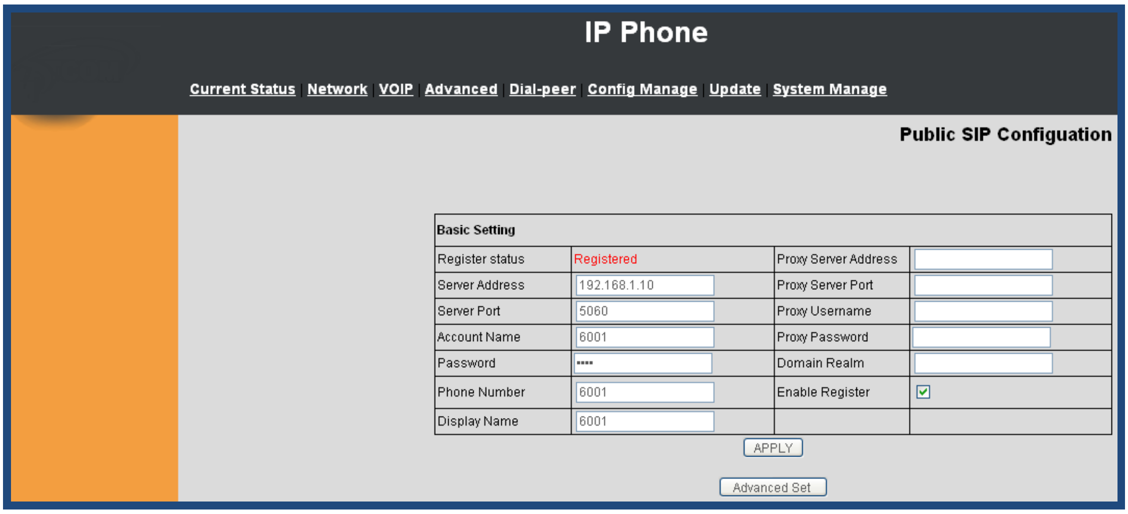


At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

Please pay attention to the **Technology** component, there is an **Analog Station** drop-down list, I choose port 1 in which port the analog phone plugs.

4.1.3 Register a SIP user 6001 in HY610

After logging into the web page of IP Phone HY610, please select VOIP option, I register the 6001 as the following illustration:



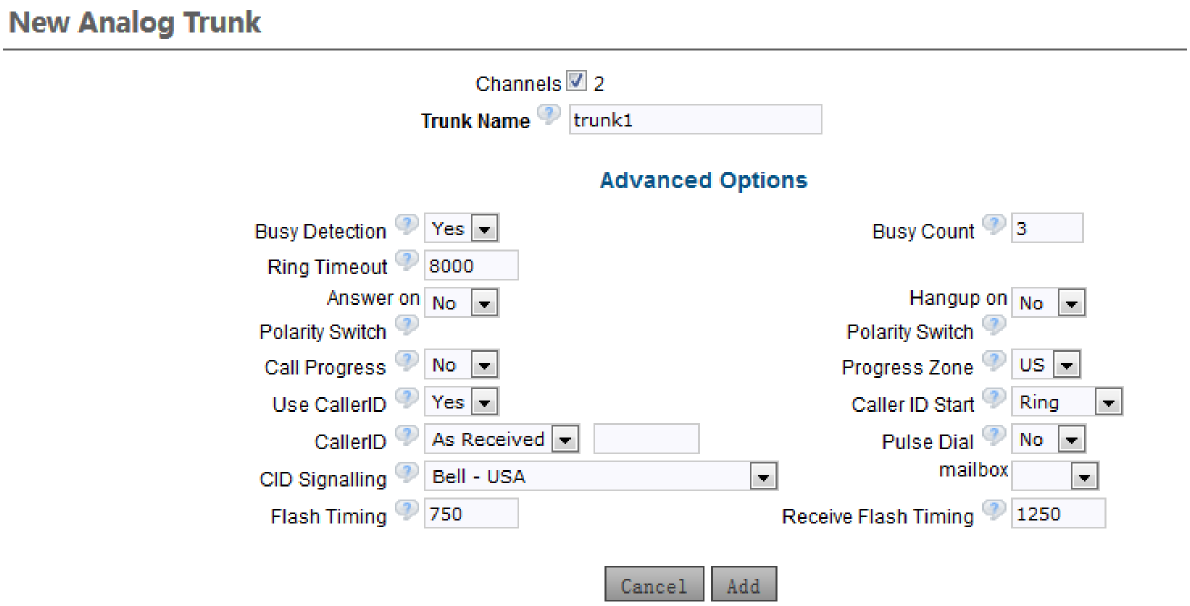
After configuring, please click on the **APPLY** button.
Now you can call each other directly between user 6001 and 6005.

4.2 How to Make a Call to Outside through PSTN

In order to dial out to PSTN with IP0x, you need an analog trunk, an outgoing calling rule, a dial plan and a user. Here I will give the simple configuration steps which show how to make a call to outside, for detail configuration, you can refer to chapter 3.

4.2.1 Create an Analog Trunk

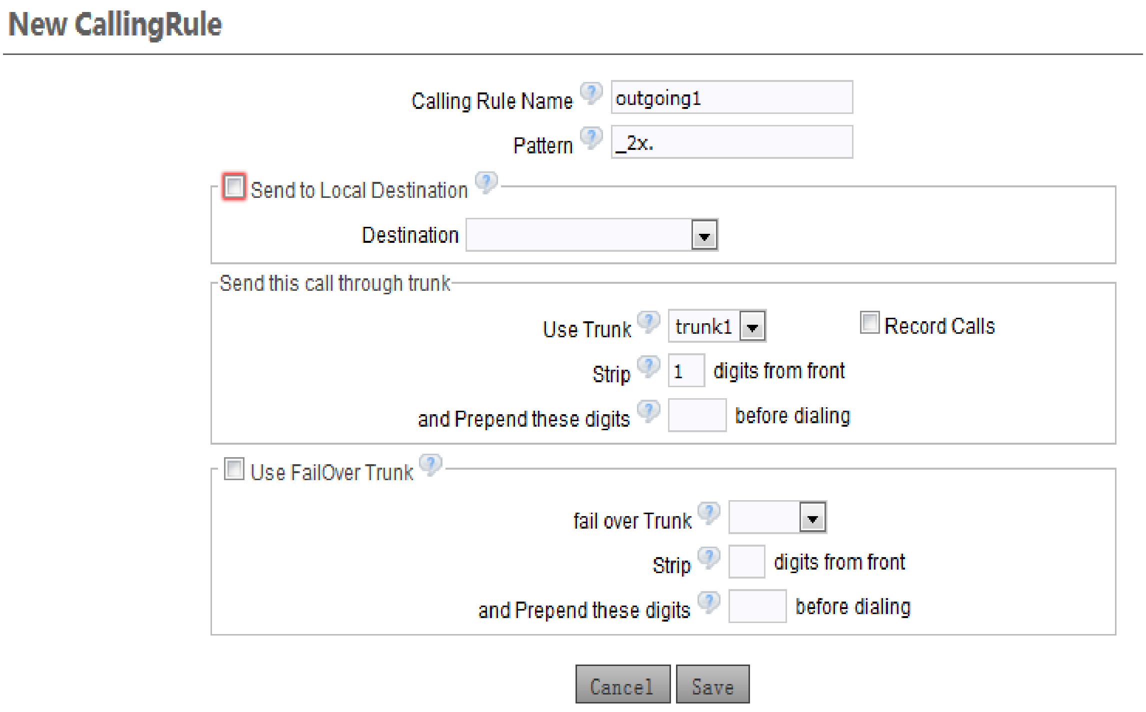
After logging into the web page of IP0x, please click on **Trunks→ Analog Trunks**, I configure an analog trunk like the following:



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.2 Create an Outgoing Calling Rule

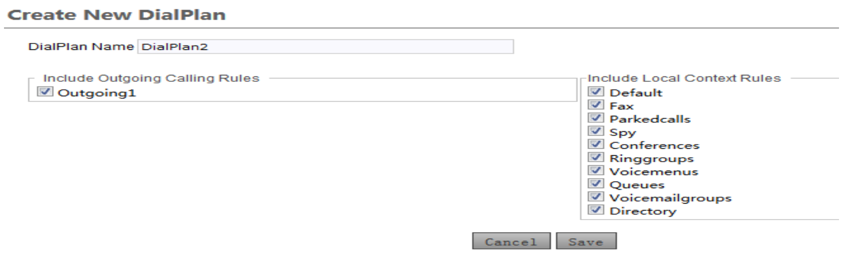
After logging into the web page of IP02, please click on **Outgoing Calling Rules→ New Calling Rule**, I configure an outgoing calling rule like the following:



At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.3 Create a Dial Plan

After logging into the web page of IP0x, please click on **Dial Plans→ New DialPlan**, I configure a dial plan like the following:



At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.4 Create a User

I will use the user 6001 I created before, here I need to reselect a dial plan for 6001, here I need to use DialPlan2, so I select DialPlan2 in the DialPlan drop-down list.
Now I can call out with prefix 2, if the caller number is 10086, I will dial 210086.

4.3 How to Get an Incoming Call from outside

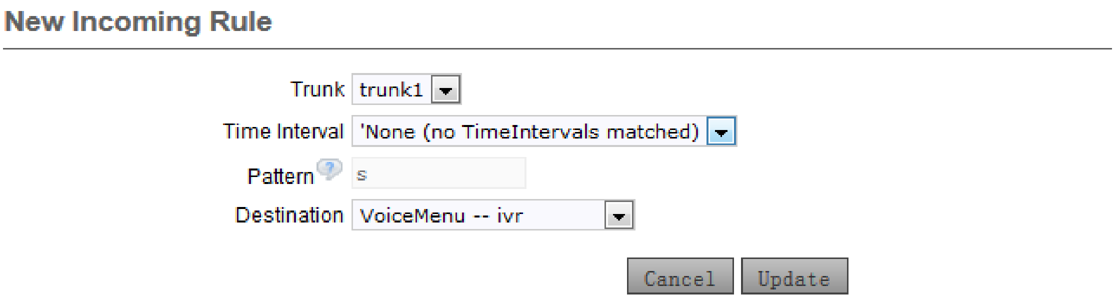
In order to get an incoming call from outside with IP0x, you need an analog trunk, an incoming calling rule, a destination (here I use IVR). Here I will give the simple configuration steps which show how to get an incoming call from outside, for detail configuration, you can refer to chapter 3.

4.3.1 Create an Analog Trunk

I use the trunk1 I created in 4.2.1

4.3.2 Create an Incoming Calling Rule

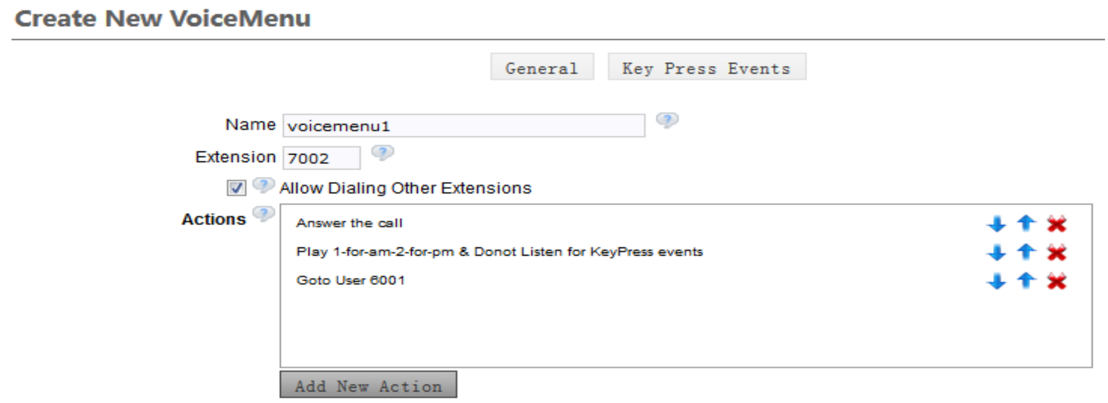
After logging into the web page of IP0x, please click on **Incoming Calling Rules→ New Incoming Rule**, I configure an incoming calling rule like the following:

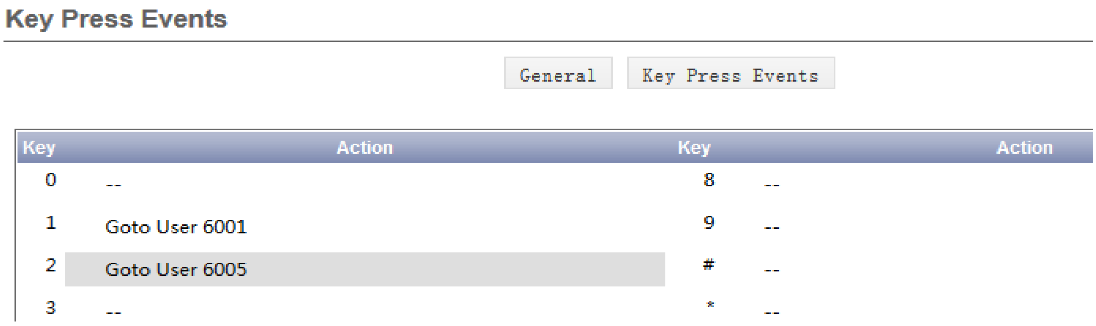


At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

4.3.3 Create a Voice Menu

After logging into the web page of IP0x, please click on **Voice Menus→ Create New VoiceMenu**, I create a voice menu like the following:





When the call comes from port 2, the system will play a record sound file, if the caller presses 1, user 6001 will ring, if the caller presses 2, user 6005 will ring. If the caller does not press any key, the call will go to 6001.

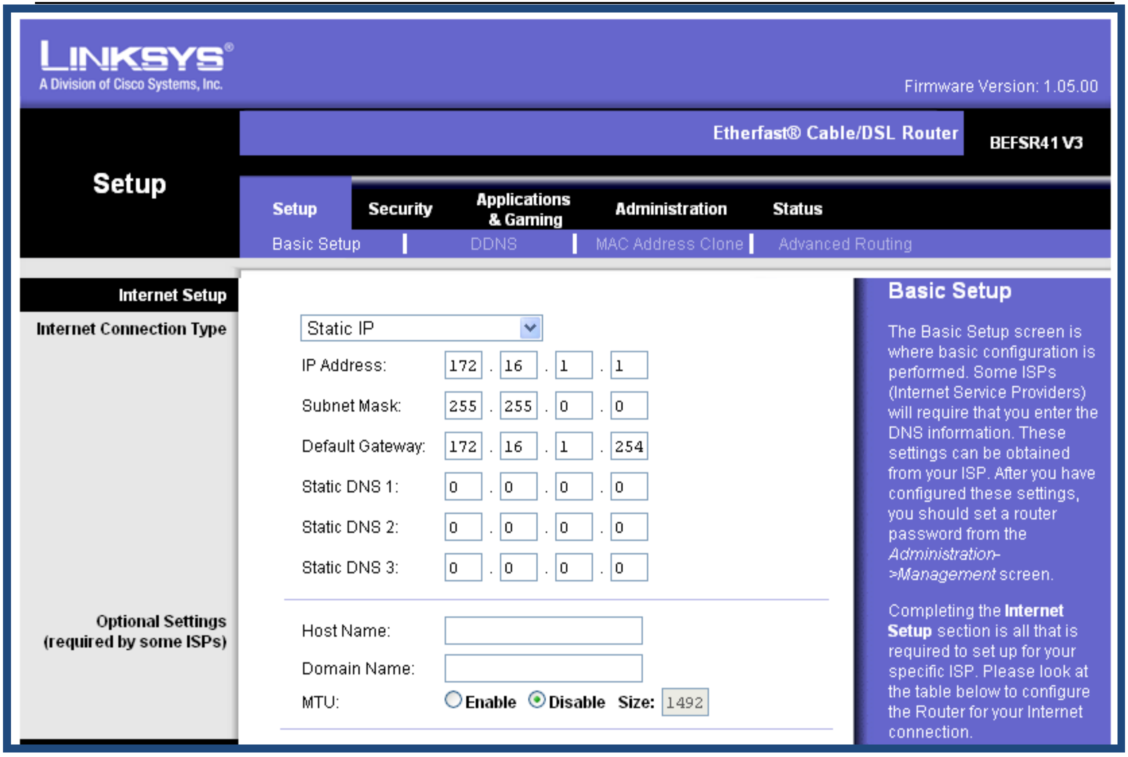
You can also configure IP0x to let 6030 call outside and get incoming call by IP0x, the steps are the same as IP0x, you can refer to configuration of IP0x.

4.4 How to Call Each Other Directly from Different Network Segment.

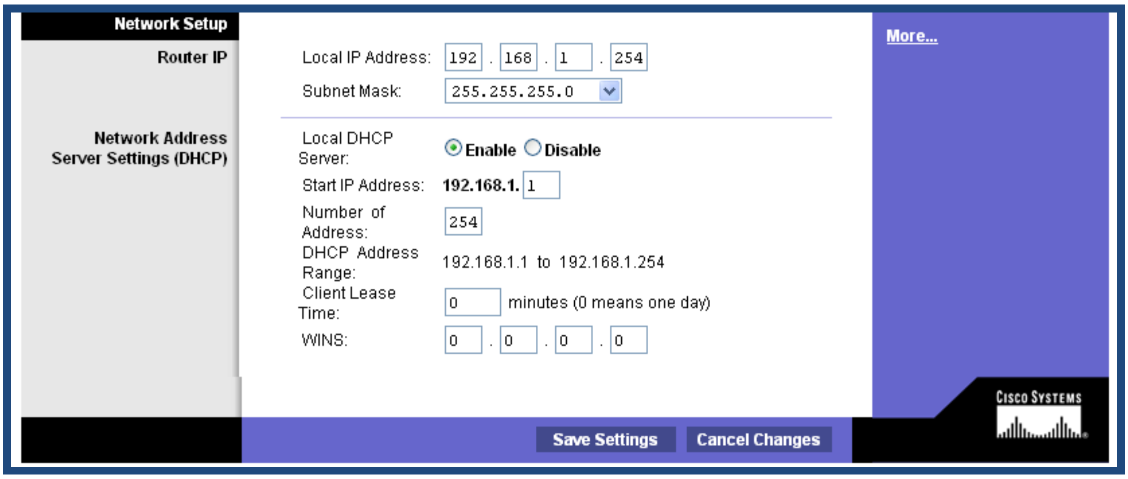
Take the user 6020, 6005 and 6001 for example, I will configure router, users and IP02, then the three users can call each other directly.

1) Set up router

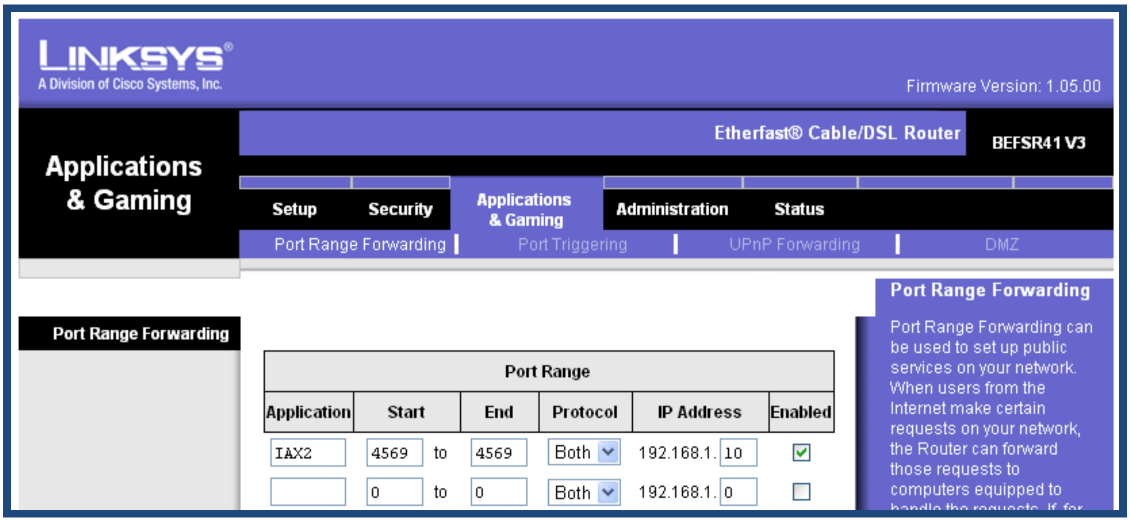
   From the web page of your router, please configure the IP address, subnet mask and default gateway of WAN port, I configured a static IP Address 172.16.1.1; Subnet Mask: 255.255.0.0; Default Gateway: 172.16.1.254. You can refer to the following:



From the web page of your router, please configure the IP address, subnet mask and DHCP, I configure them like the following:



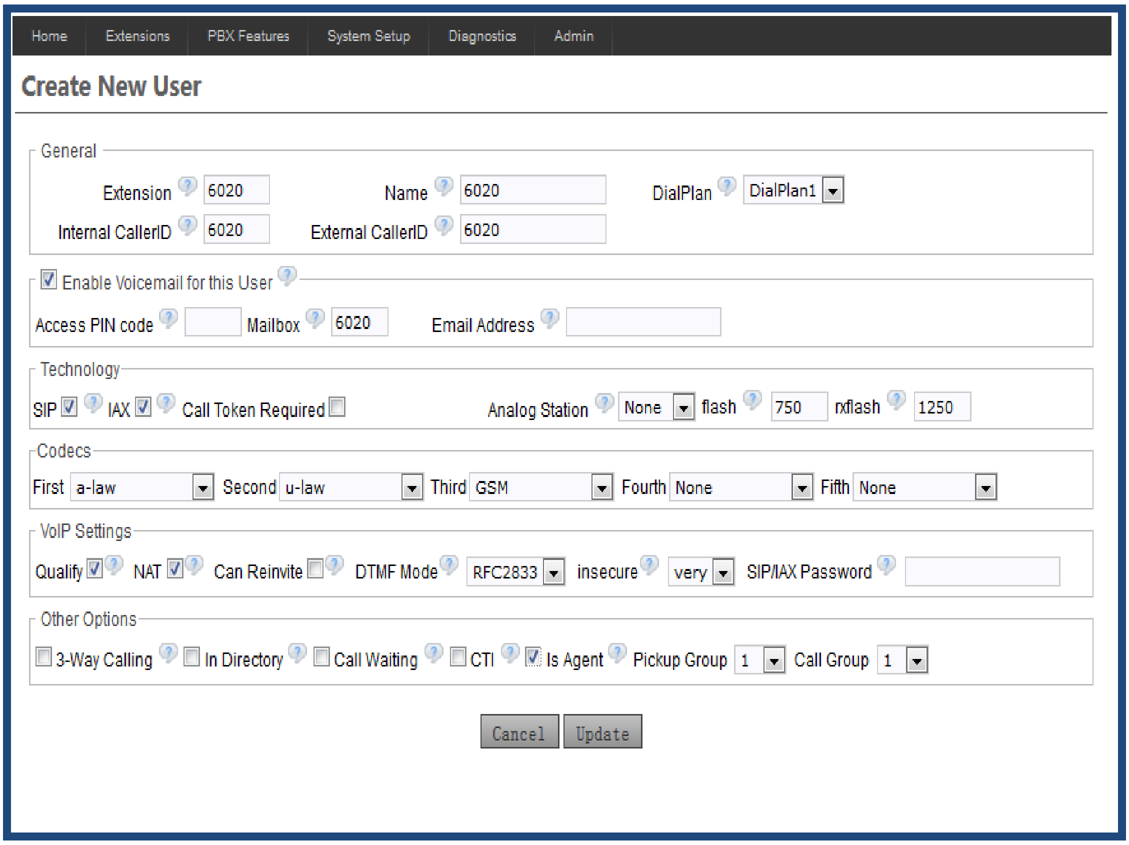
From the webpage of your router, please configure port range forwarding like the following:



The user 6020 uses IAX2, the port number is 4569, 192.168.1.10 is the IP address of IP0x.

2) Add an IAX user 6020 in IP0x

After logging into the web page of IP0x, please click on **Users→ Create New User**, I configure 6020 like the following:

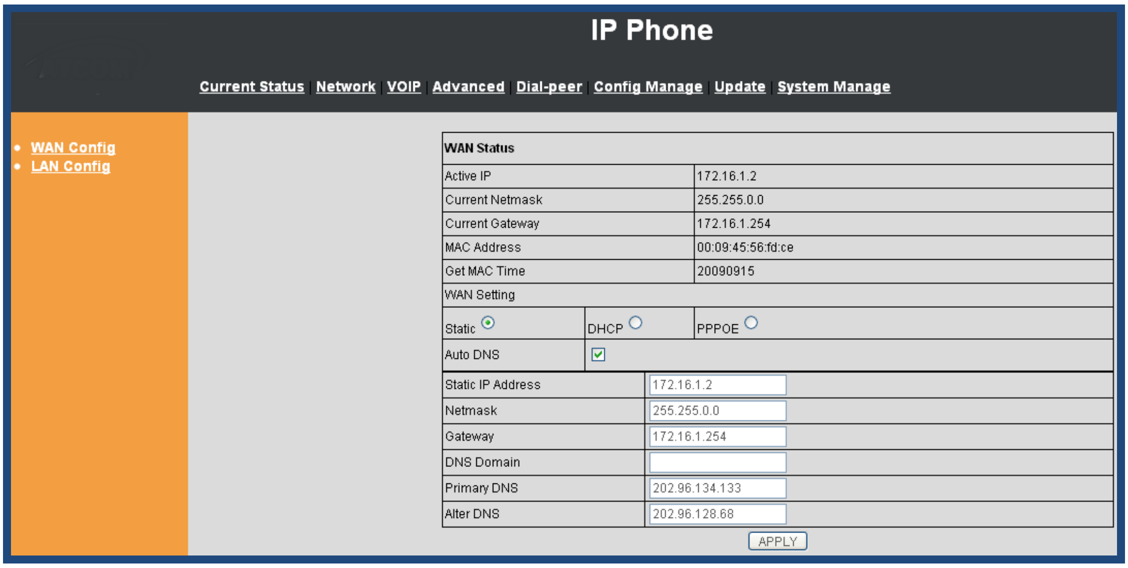


At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

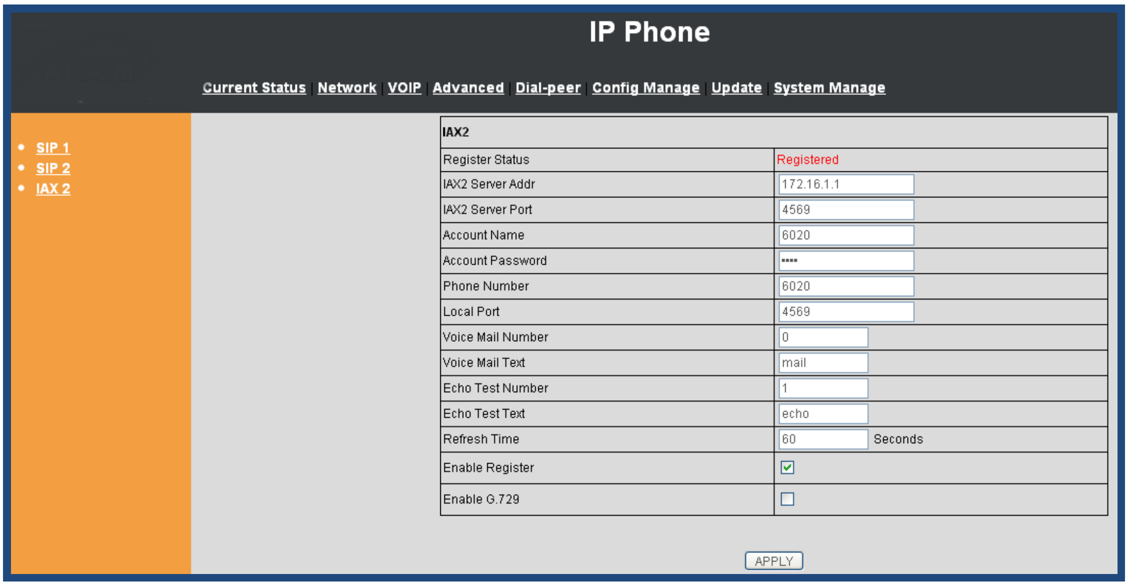
3) Set up HY-620 and register an IAX2 user 6020

After logging into web page of IP Phone HY-620, please select **Network** option to enter the screen of configuring IP Address. I set up a static IP Address: 172.16.1.2; Netmask: 255.255.0.0; Gateway: 172.16.1.254. After finishing the configuration, please click on the **Apply** button. You

can refer to the following screen:



Please select the **VOIP** option, then select the **IAX2** option, I register the IAX2 user 6020 as the following illustration:



After configuring, please click on the **APPLY** button.

Attention: here you must register IAX2 user instead of SIP user, because the user 6020 is not in the same network segment as IP0x. If you use SIP user, you can not get sound when the communication is established.

Now you can call each other among 6020,6001 and 6005 directly.

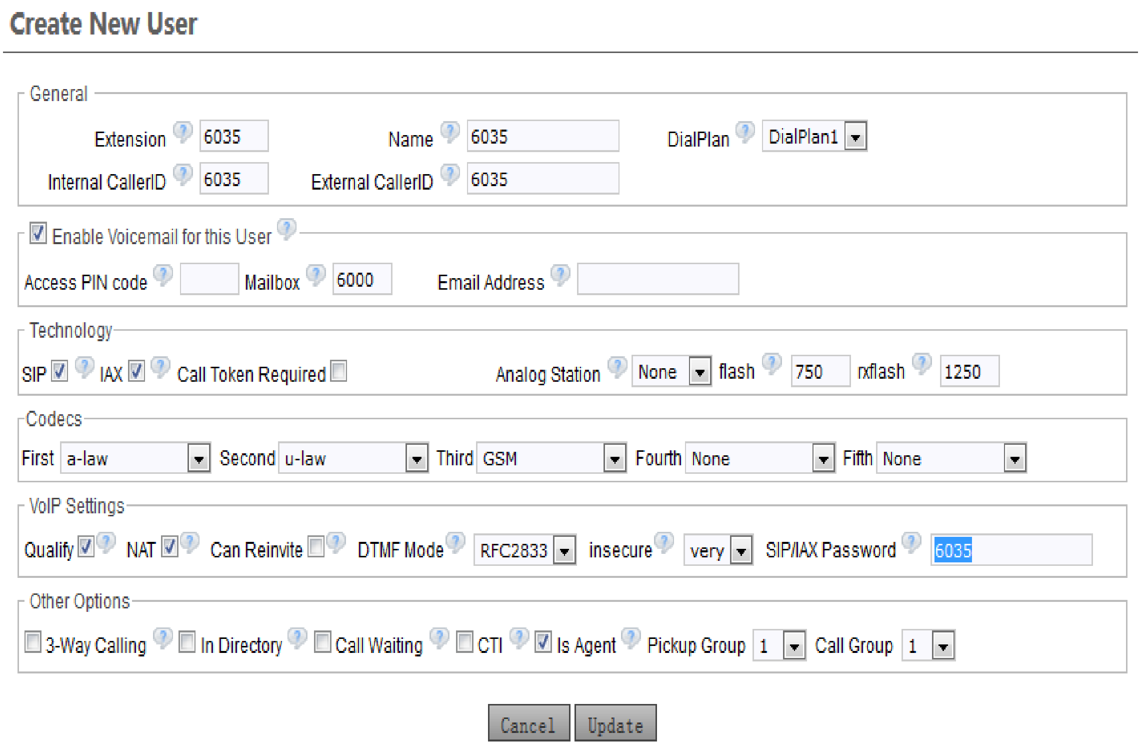
4.5 How to Call through VoIP Trunk

4.5.1 Call from IP02 to IP08

In order to call from IP02 to IP08, I will create a SIP user in IP08 for the SIP trunk in IP02, create

a SIP trunk, an outgoing call rule and a dial plan in IP02.

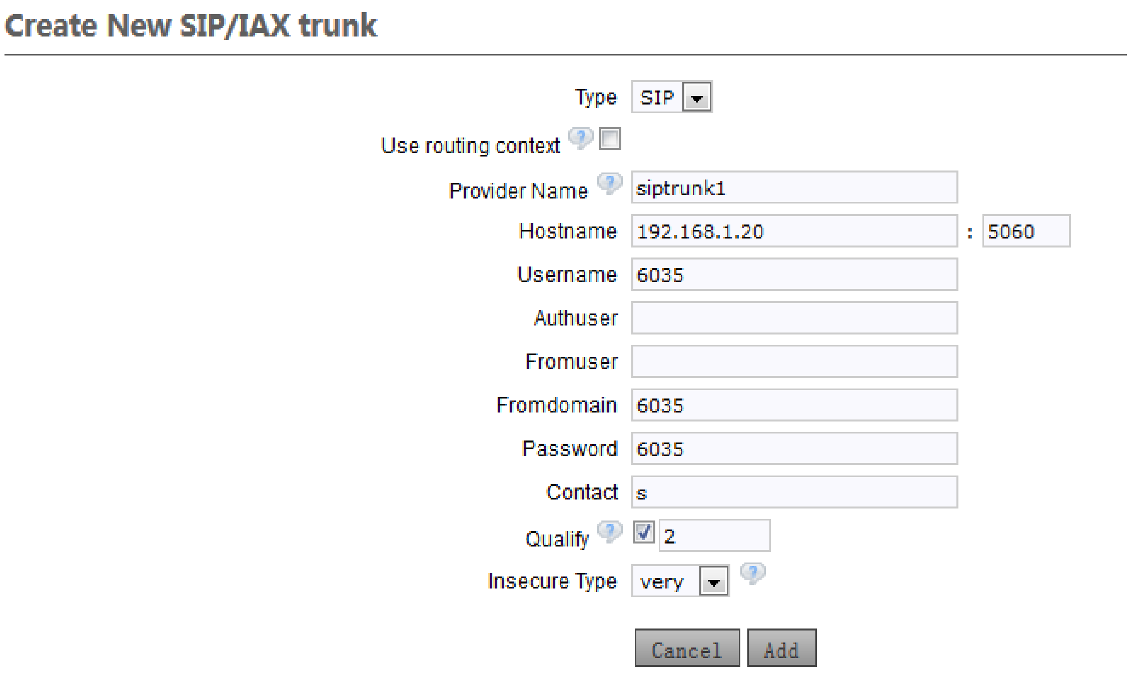
1) Add an SIP user 6035(it will be used as SIP trunk in IP02) in IP08, after logging into the web page of IP08, please click on **Users→ Create New User**, I add the user 6035 like the following:



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

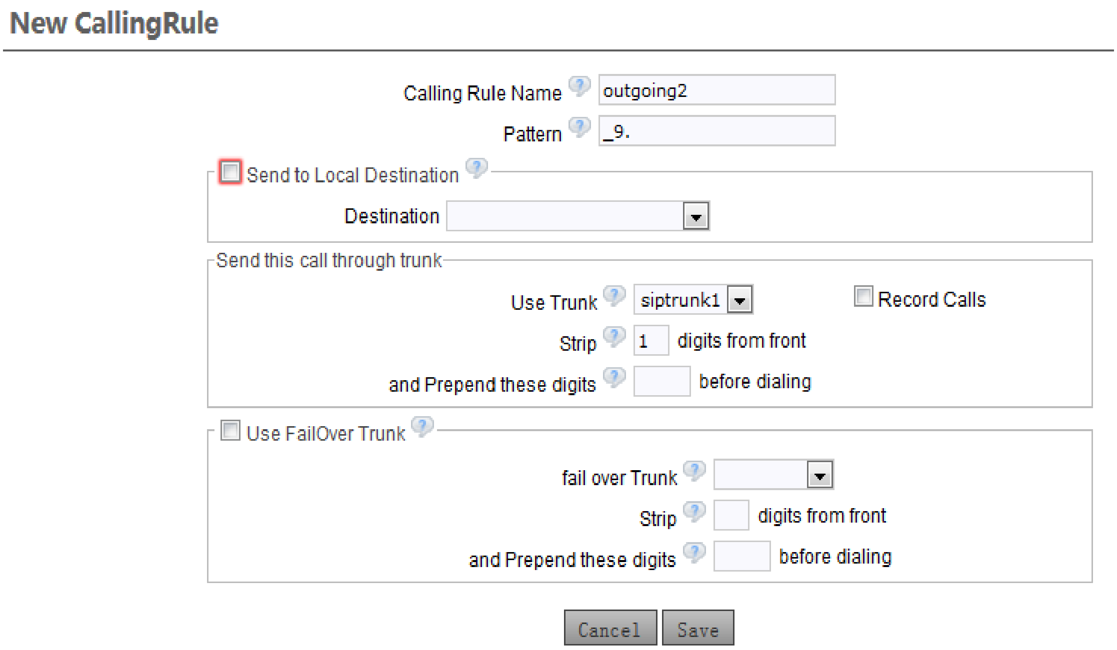
Add a SIP user 6030 in IP08 for HY620, the way is the same as adding 6035.

2) Add a VoIP trunk in IP02, after logging into the webpage of IP02, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIP trunk1 like the following:



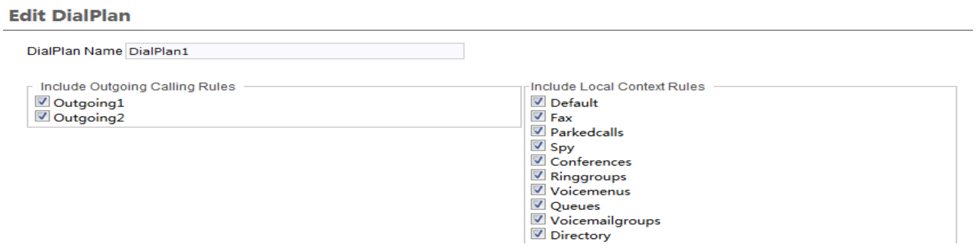
After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page.

3) Create an outgoing calling rule in IP02, after logging into the webpage of IP02, please click on **Outgoing Calling Rules→New Calling Rule**, I configure an outgoing2 rule like the following:



After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4) Create a dial plan in IP02, after logging into the webpage of IP02, please click on **Dial Plans→New DialPlan**, I configure a dialplan2 like the following:



After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

In configuration screens of 6001 and 6005, please select dialplan1 in the **DialPlan** drop-down list

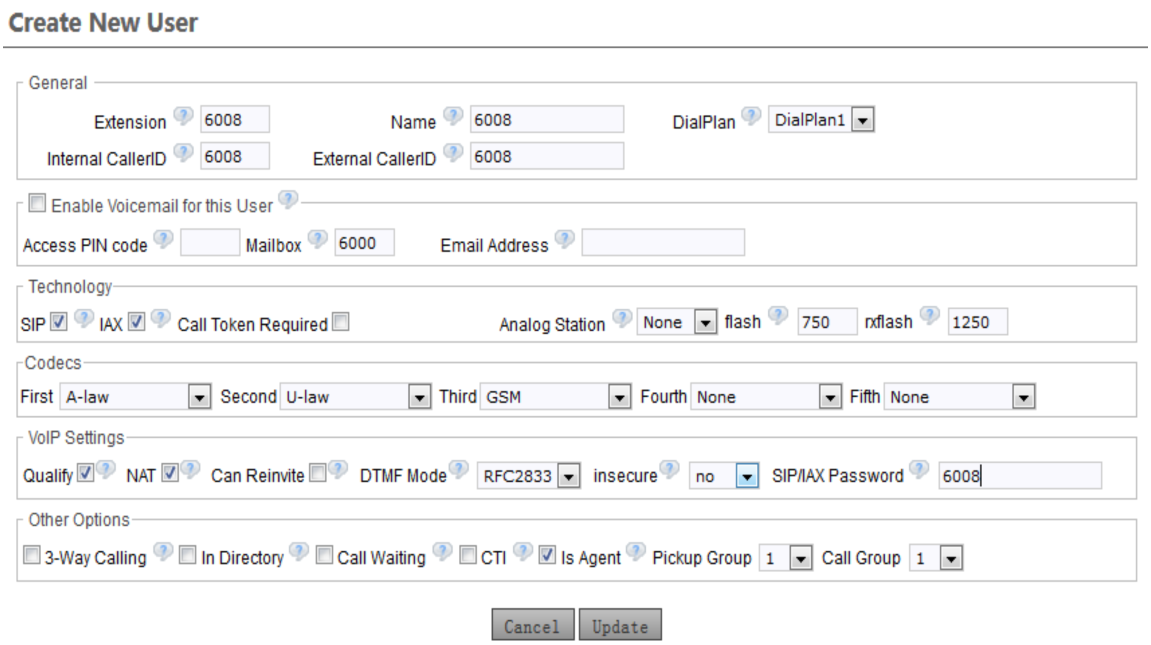
Now you can call from 6001 and 6005 to 6030 by dialing 96030

4.5.2 Call from IP08 to IP02

In order to call from IP08 to IP02, I will create a SIP user in IP02 for the SIP trunk in IP08, create a SIP trunk, an outgoing call rule and a dial plan in IP08.

1） Add a user 6008 in IP02

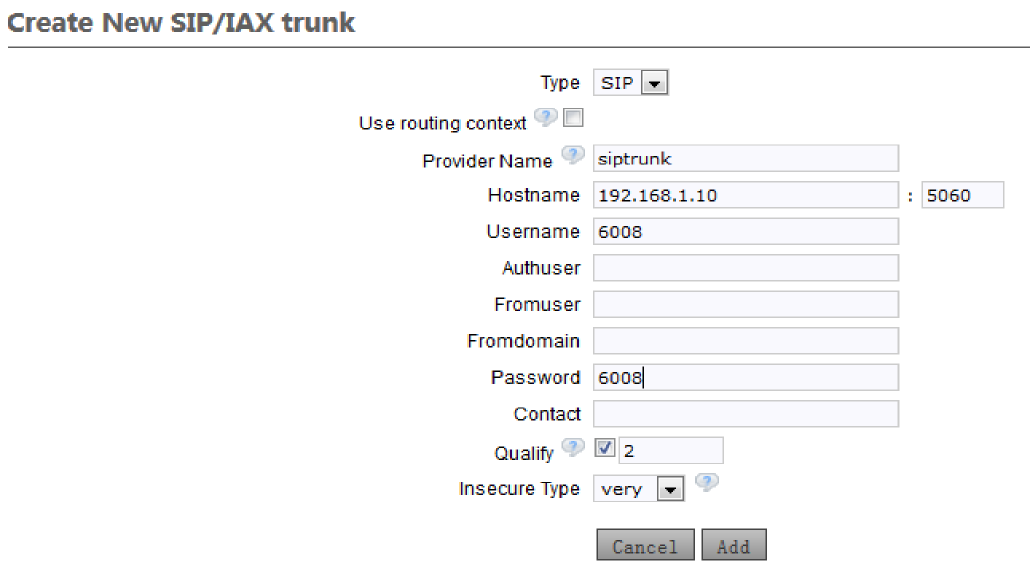
Add a SIP user: 6008, after logging into the web page of IP02, please click on **Users→ Create New User**, I add a user 6008 like the following:



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

2) Create a SIP trunk in IP08

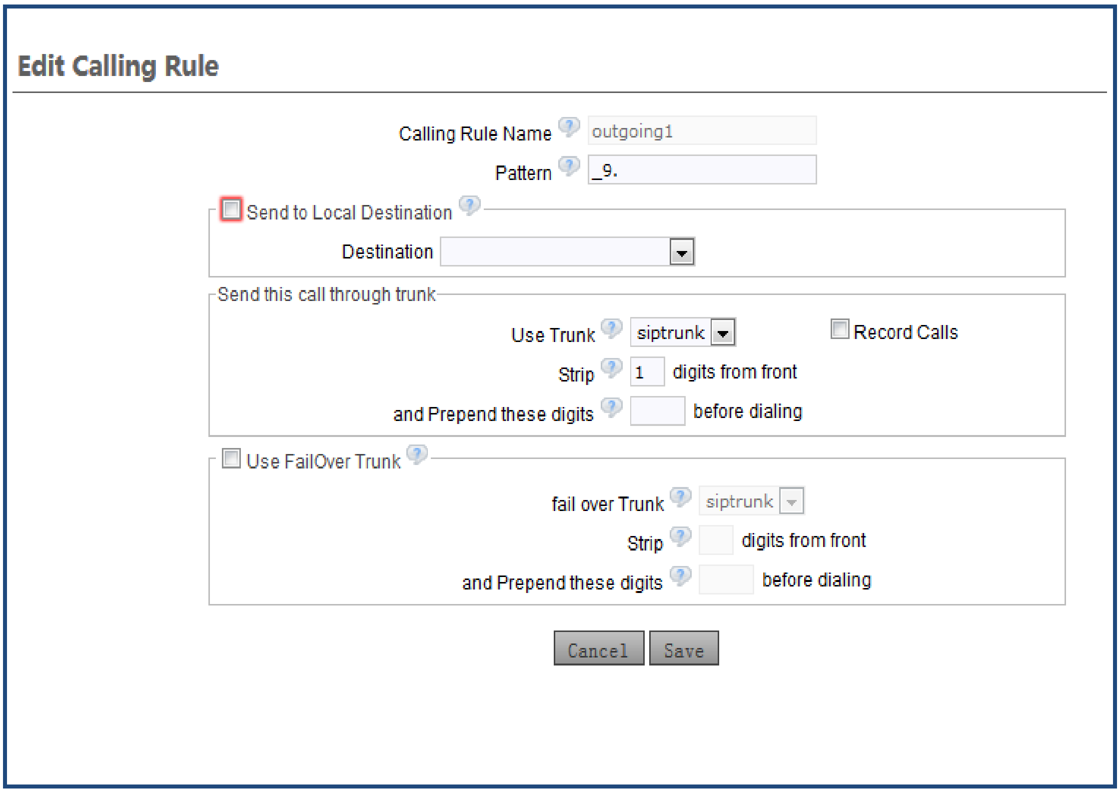
Add a VoIP trunk in IP08, after logging into the webpage of IP08, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIP trunk like the following:



After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page.

3) Create an outgoing calling rule in IP08

After logging into the webpage of IP08, please click on **Outgoing Calling Rules→New Calling Rule**, I configure an outgoing1 rule like the following:



After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4) Create a dial plan in IP08

After logging into the webpage of IP08, please click on **Dial Plans→New DialPlan**, I configure a dialplan1 like the following:

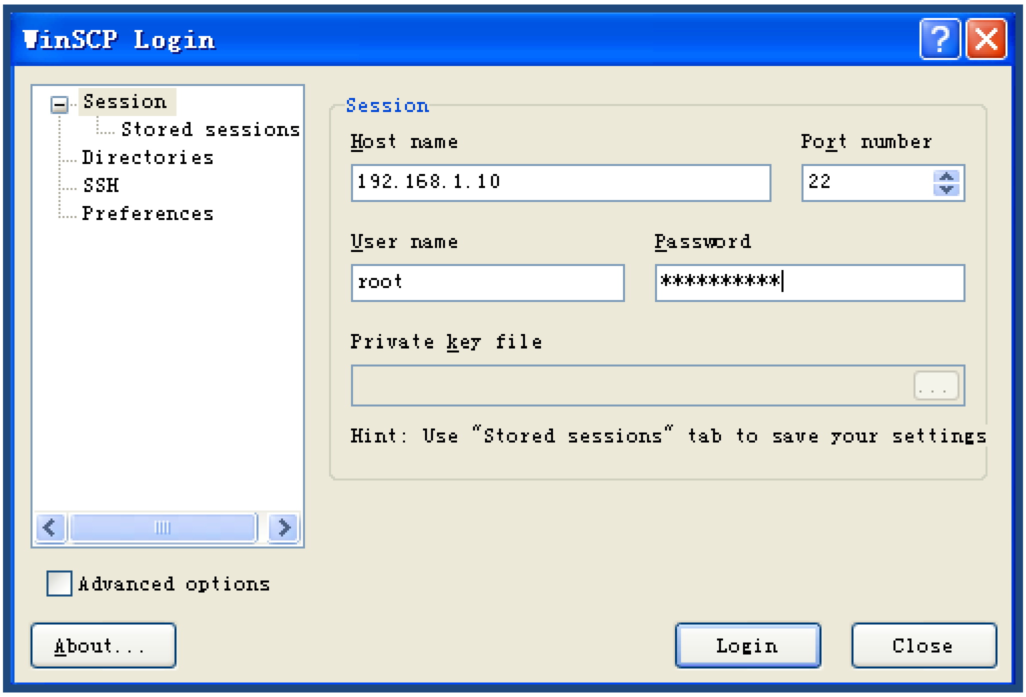
After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

In configuration screens of 6030, please select dialplan1 in the **DialPlan** drop-down list.

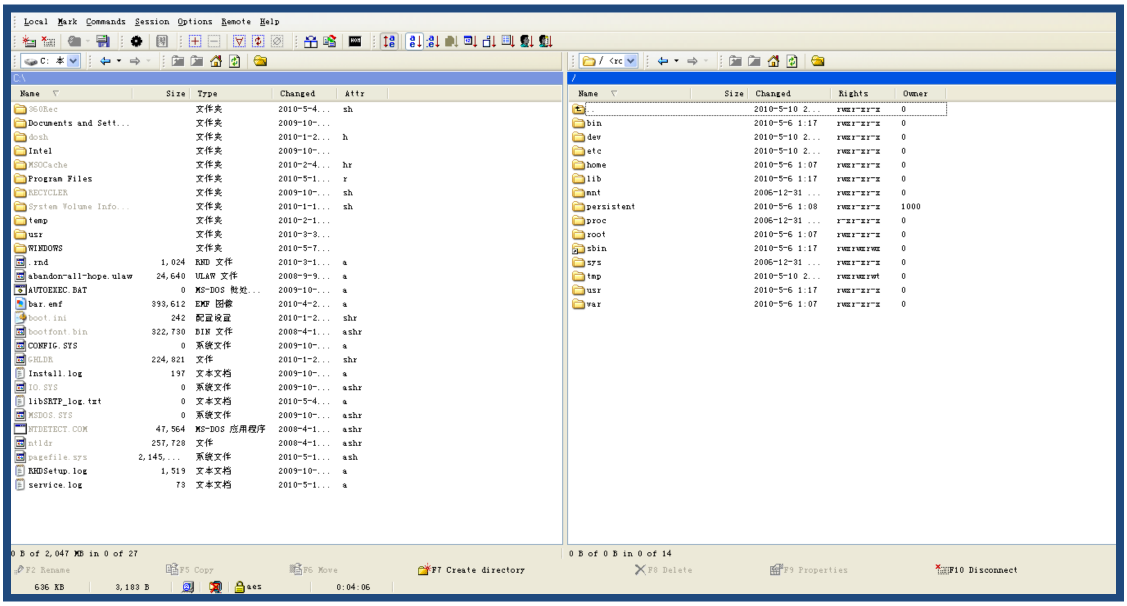
Now you can call from 6030 to 6001 and 6005 by dialing with prefix 9.

4.6 How to Transfer Files between Windows PC and IP0x

Using WinSCP software, it is the most convenient way to transfer files between windows PC and IP0x. Open your WinSCP software，enter the IP Address, username, password of IP0x like the following screen:

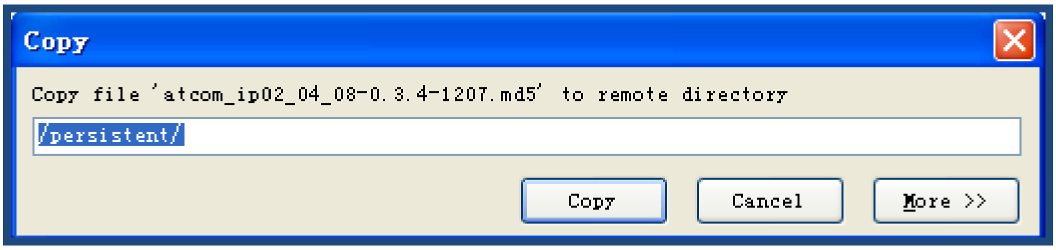


At last, click on **Login** button, then you can get the following screen:



The left part of the screen displays directories and files of your windows PC, the right part of the screen displays directories of IP0x.

If you want to transfer a file from windows PC to IP0x, you just need to choose the file and drag it to the directory of IP0x, at last, click on **copy** button in the popping-up screen like the following:



Chapter 5 Reference

http://www.houyuanhk.com

http://www.globalsources.com/houyuanhk.co